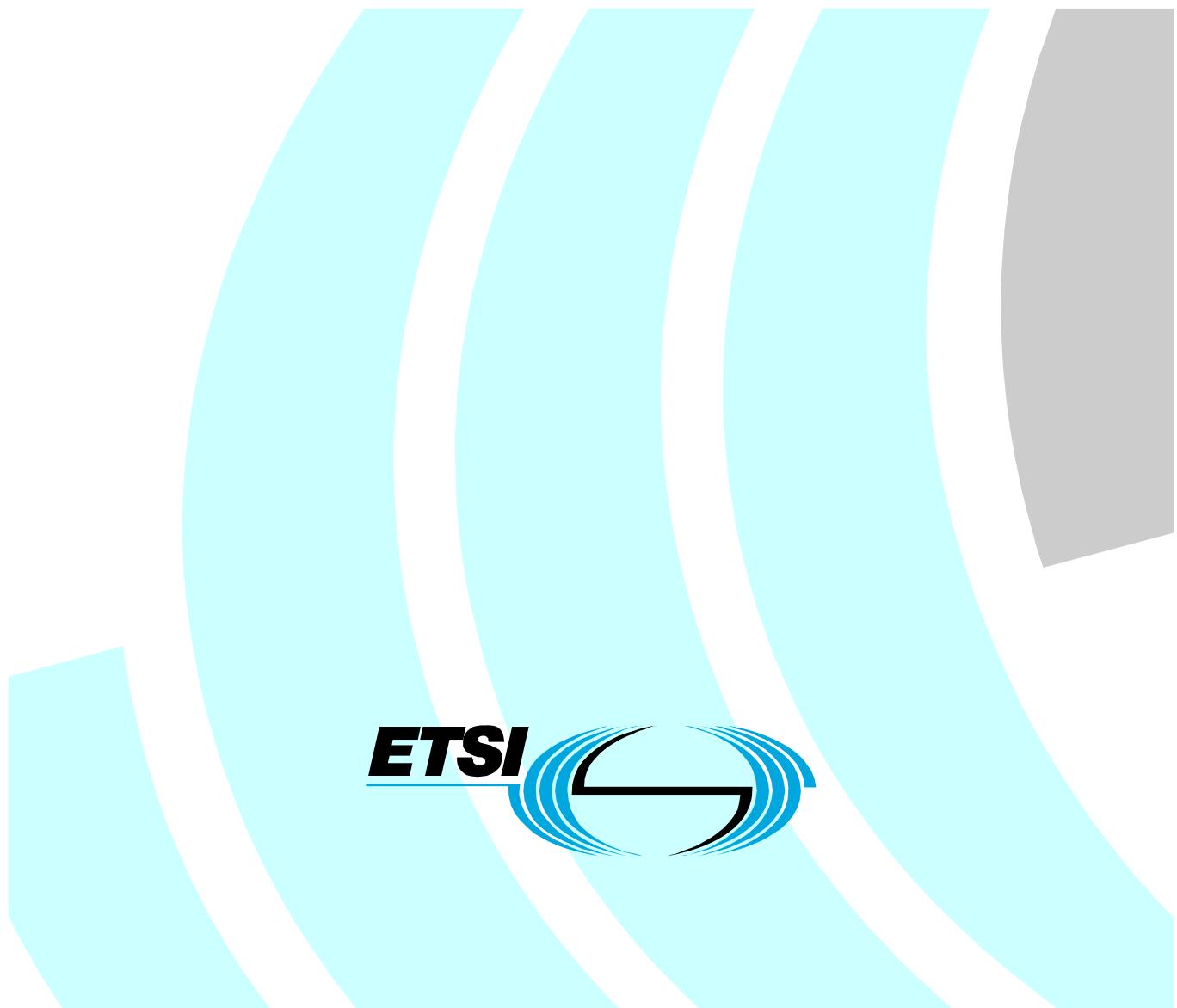


**Telecommunications and Internet Converged Services and
Protocols for Advanced Networking (TISPAN);
Interworking between Session Initiation Protocol (SIP) and
Bearer Independent Call Control Protocol (BICC) or
ISDN User Part (ISUP);
Part 2: Test Suite Structure and Test Purposes (TSS&TP)
for Profile A and B**



Reference

RTS/TISPAN-06028-2-NGN

Keywords

BICC, CTS, interworking, SIP, testing, TSS&TP

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN).

The present document is part 2 of a multi-part deliverable covering the Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol (BICC) or ISDN User Part (ISUP), as identified below:

- Part 1: "Protocol Implementation Conformance Statement (PICS)";
- Part 2: "Test Suite Structure and Test Purposes (TSS&TP) for Profile A and B";**
- Part 3: "Test Suite Structure and Test Purposes (TSS&TP) for Profile C";
- Part 4: "Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) for Profile A and B";
- Part 5: "Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) for Profile C".

1 Scope

The present document specifies the Test Suite Structure and Test Purposes (TSS&TP) for the Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control Protocol or ISDN User Part for the **Profile A and Profile B** described in the ITU-T Recommendation Q.1912.5 [1] and EN 383 001 [2].

A further part of the present document specifies the Abstract Test Suite (ATS) and partial Protocol Implementation eXtra Information for Testing (PIXIT) proforma based on the present document.

2 References

References are either specific (identified by date of publication and/or edition number or version number) or non-specific.

- For a specific reference, subsequent revisions do not apply.
- Non-specific reference may be made only to a complete document or a part thereof and only in the following cases:
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 - for informative references.

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NOTE: While any hyperlinks included in this clause were valid at the time of publication ETSI cannot guarantee their long term validity.

2.1 Normative references

The following referenced documents are indispensable for the application of the present document. For dated references, only the edition cited applies. For non-specific references, the latest edition of the referenced document (including any amendments) applies.

- [1] ITU-T Recommendation Q.1912.5: "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".
- [2] ETSI EN 383 001: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control (BICC) Protocol or ISDN User Part (ISUP) [ITU-T Recommendation Q.1912.5, modified]".
- [3] ITU-T Recommendations Q.761 to Q.764 (2000): "Signalling System No.7 ISDN User Part (ISUP)".
- [4] ITU-T Recommendations Q.1902.1 to Q.1902.4 (2001): "Bearer Independent Call Control Protocol (BICC)".
- [5] ITU-T Recommendation Q.850 (1998): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN User Part".
- [6] IETF RFC 3261 (2002): "SIP: Session Initiation Protocol".

- [7] IETF RFC 3312 (2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".
- [8] ISO/IEC 9646-1 (1994): "Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 1: General Concepts".
- [9] ISO/IEC 9646-3 (1992): " Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 3: The Tree and Tabular Combined Notation (TTCN)".
- [10] ISO/IEC 9646-7 (1995): " Information technology - Open Systems Interconnection - Conformance testing methodology and framework - Part 7: Implementation Conformance Statement".
- [11] ITU-T Recommendation E.164: "The international public telecommunication numbering plan".
- [12] IETF RFC 3267: "Real-Time Transport Protocol (RTP) Payload Format and File Storage Format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) Audio Codecs".
- [13] ITU-T Recommendation Q.939: "Typical DSS 1 service indicator codings for ISDN telecommunications services".
- [14] ETSI ES 283 027: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Endorsement of the SIP-ISUP Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks [3GPP TS 29.163 (Release 7), modified]".
- [15] ETSI TS 183 008: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); PSTN/ISDN simulation services Terminating Identification Presentation (TIP) and Terminating Identification Restriction (TIR); Protocol specification".

2.2 Informative references

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area. For non-specific references, the latest version of the referenced document (including any amendments) applies.

Not applicable.

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in SIP/ISUP interworking reference specification, in ISDN layer 3 reference specification, in ISO/IEC 9646-1 [8], in ISO/IEC 9646-3 [9], in ISO/IEC 9646-7 [10] and the following apply:

Abstract Test Case (ATC): complete and independent specification of the actions required to achieve a specific test purpose, defined at the level of abstraction of a particular Abstract Test Method, starting in a stable testing state and ending in a stable testing state

Abstract Test Method (ATM): description of how an SUT is to be tested, given at an appropriate level of abstraction to make the description independent of any particular realization of a Means of Testing, but with enough detail to enable abstract test cases to be specified for this method

Abstract Test Suite (ATS): test suite composed of abstract test cases

Implementation Under Test (IUT): implementation of one or more OSI protocols in an adjacent user/provider relationship, being part of a real open system which is to be studied by testing

Means of Testing (MOT): combination of equipment and procedures that can perform the derivation, selection, parameterization and execution of test cases, in conformance with a reference standardized ATS, and can produce a conformance log

PICS proforma: document, in the form of a questionnaire, which when completed for an implementation or system becomes the PICS

PIXIT proforma: document, in the form of a questionnaire, which when completed for the SUT becomes the PIXIT

Point of Control and Observation (PCO): point within a testing environment where the occurrence of test events is to be controlled and observed, as defined in an Abstract Test Method

Pre-test condition: setting or state in the SUT which cannot be achieved by providing stimulus from the test environment

Protocol Implementation Conformance Statement (PICS): statement made by the supplier of a protocol claimed to conform to a given specification, stating which capabilities have been implemented

Protocol Implementation eXtra Information for Testing (PIXIT): statement made by a supplier or implementor of an SUT (protocol) which contains or references all of the information related to the SUT and its testing environment, which will enable the test laboratory to run an appropriate test suite against the SUT

SIP number: number conforming to the numbering and structure specified in ITU-T Recommendation E.164 [11]

System Under Test (SUT): real open system in which the SUT resides

User: access protocol entity at the User side of the user-network interface where a T reference point or coincident S and T reference point applies

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ATC	Abstract Test Case
ATM	Abstract Test Method
ATP	Access Transport Parameter
ATS	Abstract Test Suite
BCI	Backward Call Indicators
CPS	Calling Party's Category
DSS1	Digital Subscriber System No. 1
FCI	Forward Call Indicators
HLC	High Layer Compatibility
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
IUT	Implementation Under Test
MOT	Means Of Testing
NCI	Nature of Connection Indicators
OBCI	Optional Backward Call Indicators
PICS	Protocol Implementation Conformance Statement
PIXIT	Protocol Implementation eXtra Information for Testing
SUT	System Under Test
TMR	Transmission Medium Requirement
TP	Test Purpose
TSS	Test Suite Structure
TTCN	Tree and Tabular Combined Notation

NOTE: The ISUP message acronyms can be found in table 2/ITU-T Recommendation Q.762 [3].

4 Implementation under test and test methods

4.1 Identification of the system and implementation under test

4.1.1 Profile A

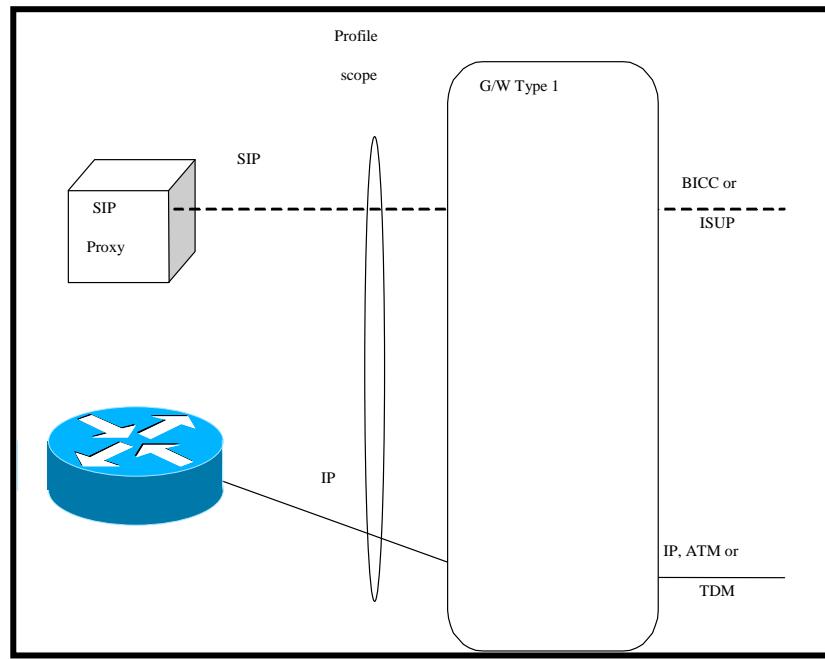


Figure 1: Profile Scope for SIP Interworking with BICC/ISUP with a Type 1 Gateway

4.1.2 Profile B

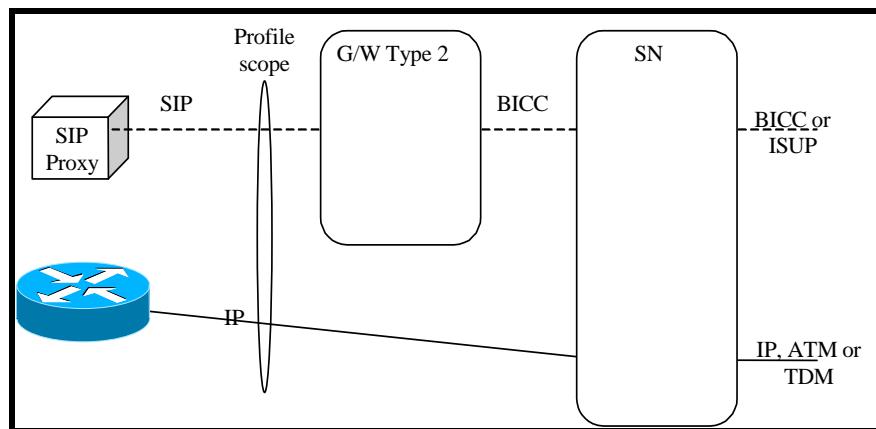


Figure 2: Profile Scope for SIP Interworking with BICC/ISUP with a Type 2 Gateway

5 Test Suite Structure (TSS)

The Test Suite Structure is in close alignment with ITU-T Recommendation Q.1912.5 [1] and EN 383 001 [2].

5.1 Interworking from SIP to ISUP (outgoing call)

SIP -ISUP Basic call		
	Sending of the Initial address message (IAM)	101xxx
	Sending of the Subsequent address message (SAM)	102xxx
	Sending of COT	103xxx
	Receipt of the Address complete message (ACM)	104xxx
	Receipt of the Call progress message (CPG)	105xxx
	Receipt of the answer message (ANM)	106xxx
	Receipt of the Connect message (CON)	107xxx
	Receipt of the Release message (REL)	108xxx
	Receipt of the BYE, CANCEL message / sending of a REL message	109xxx
	Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	1010xxx
	Receipt of the SUSPEND Message (SUS)	1011xxx
	Receipt of the RESUME Message (RES)	1012xxx

Figure 3: Basic call -
Test suite structure for interworking between SIP to ISUP (outgoing call)

5.2 Interworking from ISUP to SIP (incoming call)

ISUP-SIP Basic call		
	Sending of the INVITE message	301xxx
	Receipt of the Subsequent address message (SAM)	302xxx
	Sending of the Address complete message (ACM)	303xxx
	Sending of the Call progress message (CPG)	304xxx
	Sending of the answer message (ANM)	305xxx
	Sending of the Connect message (CON)	306xxx
	Receipt of the Release message (REL)	307xxx
	Sending of the Release Message (REL)	308xxx
	Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	309xxx

Figure 4: Basic call -
Test suite structure for interworking between ISUP to SIP (incoming call)

5.3 Supplementary Services - Interworking from SIP to ISUP (outgoing call)

SIP-ISUP Supplementary Services	
Calling Line Identification (CLI)	501xxx
Call Hold (HOLD)	502xxx
Terminal Portability (TP)	503xxx
Conference Calling (CONF)	504xxx
Three-Party (3PTY)	505xxx
Connected Line Identification (COL)	506xxx
Malicious call identification (MCID)	507xxx
Subaddressing (SUB)	508xxx
Call Diversion (CDIV)	509xxx
Call Waiting (CW)	510xxx
User to User Signalling (UUS)	511xxx
Explicit Call transfer (ECT)	512xxx
Completion of Call to Busy Subscriber (CCBS)	513xxx
Completion of Calls on No reply (CCNR)	514xxx
Anonymous Call Rejection (ACR)	515xxx

**Figure 5: Supplementary Services -
Test suite structure for interworking between SIP to ISUP (outgoing call)**

5.4 Supplementary Services - Interworking from ISUP to SIP (incoming call)

ISUP-SIP	
Calling Line Identification (CLI)	601xxx
Call Hold (HOLD)	602xxx
Terminal Portability (TP)	603xxx
Conference Calling (CONF)	604xxx
Three-Party (3PTY)	605xxx
Connected Line Identification (COL)	606xxx
Subaddressing (SUB)	607xxx
Closed User Group (CUG)	608xxx
Call Diversion (CDIV)	609xxx
Call Waiting (CW)	FFS
User to User Signalling (UUS)	610xxx
Explicit Call transfer (ECT)	611xxx
Completion of Calls on No reply (CCNR)	FFS
Completion of Call to Busy Subscriber (CCBS)	FFS

**Figure 6: Supplementary Services -
Test suite structure for interworking between ISUP to SIP (outgoing call)**

6 Test purposes (TP)

6.1 Introduction

For each test requirement a Test Purpose (TP) is defined.

6.1.1 Test purpose (TP) naming convention

For each test requirement a Test Purpose (TP) is defined.

All test purposes belong to the main group ISUP_SIP_Interworking. Groups are organized according to the test suite structure (TSS). Each test purpose is presented in a separate table. The first row of the table contains the following items:

TP	Identifier of the test purpose;
SIP reference	the reference to the requirement in the DSS1 layer 3 Recommendation, which led to the TP;
ISUP reference	the reference to the requirement in the interworking specification and the requirement in the SIP-UP Recommendation, which led to the TP.

6.1.2 Source of test purpose definition

The test purposes have been developed based on ITU-T Rec Q.1912.5 [1] and EN 383 001 [2].

6.1.3 Test purpose structure

The test purpose structure is according to the test suite structure (TSS).

6.2 Test purposes for the basic call

6.2.1 Interworking from SIP to ISUP (Outgoing Call)

6.2.1.1 Sending of the Initial Address Message (IAM)

TP101001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.1 (1, a)
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	NOT PICS 4/4 AND PICS 4/5	
ISUP selection criteria:		
Test purpose:	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, without an SDP offer and reliable provisional responses are supported: <ul style="list-style-type: none"> • the SUT shall immediately send an SDP offer including a media description with A-law (PCMA), but not μ-law (PCMU) within a 183 Session Progress message; • sends a IAM message upon receipt of the SDP answer with media description. 	
SIP Parameter values:	SIP: 183 SDP1; PRACK SDP2	
ISUP Parameter values:		
Comments:	SIP INVITE 183 Session Progress PRACK 200 OK PRACK <div style="text-align: center; margin-top: 10px;">   </div>	SUT ISUP/BICC IAM

TP101002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.1 (1,i,b)	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/		
SIP selection criteria:	PICS 4/4 AND PICS 4/5		
ISUP selection criteria:	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1		
Test purpose:	<p>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, without an SDP offer and reliable provisional responses are supported:</p> <ul style="list-style-type: none"> the SUT shall immediately send an SDP offer including a media description with A-law (PCMA), but not μ-law (PCMU) within a 183 Session Progress message; sends a IAM message whereby the Continuity Indicator of the Nature of Connection Indicators parameter shall be set to "<i>COT to be expected</i>". 		
SIP Parameter values:	SIP: 183 SDP1; PRACK SDP2; UPDATE SDP3; 200 OK UPDATE SDP 4		
ISUP Parameter values:	IAM; Continuity Indicator: COT to be expected COT; Continuity Indicator: continuity		
Comments:	SIP INVITE 183 Session Progress PRACK 200 OK PRACK UPDATE(SDP)	SUT → ← → ← → → COT	BICC IAM

TP101003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.1 (1,i,b)
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1	
Test purpose:	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, without an SDP offer and reliable provisional responses are supported: <ul style="list-style-type: none"> the SUT shall immediately send an SDP offer including a media description with A-law (PCMA), but not μ-law (PCMU) within a 183 Session Progress message; sends a IAM message whereby the Continuity check indicator in the Nature of Connection Indicators parameter is set to "continuity check required on this circuit". 	
SIP Parameter values:	SIP: 183 SDP1; PRACK SDP2; UPDATE SDP3; 200 OK UPDATE SDP 4	
ISUP Parameter values:	IAM; Continuity Indicator: continuity check required on this circuit COT; Continuity Indicator: continuity check successful	
Comments:	SIP INVITE → 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE(SDP) →	SUT ISUP IAM COT

TP101004	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.1 (1,i,b)
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	NOT PICS 1/6 AND NOT PICS 4/1	
Test purpose:	<p>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, without an SDP offer and reliable provisional responses are supported:</p> <ul style="list-style-type: none"> the SUT shall immediately send an SDP offer including a media description with A-law (PCMA), but not μ-law (PCMU) within a 183 Session Progress message; sending of the IAM shall be deferred until all preconditions have been met. 	
SIP Parameter values:	SIP: 183 SDP1; PRACK SDP2; UPDATE SDP3; 200 OK UPDATE SDP 4	
ISUP Parameter values:		
Comments:	SIP INVITE → 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE → 200 OK UPDATE ←	SUT → IAM

TP101005	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.1 (1,ii,a)
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/5	
ISUP selection criteria:	PICS 1/6	
Test purpose:	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, without an SDP offer and reliable provisional responses are supported: <ul style="list-style-type: none"> the SUT shall immediately send an SDP offer including a media description with both A-law (PCMA) and μ-law (PCMU) included and μ-law (PCMU) shall take precedence over A-law (PCMA) within a 183 Session Progress message; sends a IAM message upon receipt of the SDP answer with media description. 	
SIP Parameter values:	SIP: 183 SDP1; PRACK SDP2	
ISUP Parameter values:	IAM;	
Comments:	SIP INVITE → 183 Session Progress ← PRACK → 200 OK PRACK →	
	→ IAM	

TP101006	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.1 (1,ii,b)
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	PICS 1/4 AND PICS 1/6 AND PICS 4/1	
Test purpose:	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, without an SDP offer and reliable provisional responses are supported: <ul style="list-style-type: none"> • the SUT shall immediately send an SDP offer including a media description with both A-law (PCMA) and μ-law (PCMU) included and μ-law (PCMU) shall take precedence over A-law (PCMA) within a 183 Session Progress message; • sends a IAM message whereby the Continuity indicator of the Nature of Connection Indicators parameter shall be set to "COT to be expected". 	
SIP Parameter values:	SIP: 183 SDP1; PRACK SDP2	
ISUP Parameter values:	IAM Continuity Indicator: COT to be expected; COT Continuity Indicator: continuity;	
Comments:	SIP INVITE → 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE →	SUT → BICC IAM COT

TP101007	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.1 (1,ii,b)
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	PICS 1/5 AND PICS 1/6 AND PICS 4/1	
Test purpose:	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, without an SDP offer and reliable provisional responses are supported: <ul style="list-style-type: none"> • the SUT shall immediately send an SDP offer including a media description with both A-law (PCMA) and μ-law (PCMU) included and μ-law (PCMU) shall take precedence over A-law (PCMA) within a 183 Session Progress message; • sends a IAM message whereby the Continuity check indicator in the Nature of Connection Indicators parameter is set to "continuity check required on this circuit". 	
SIP Parameter values:	SIP: 183 SDP1; PRACK SDP2	
ISUP Parameter values:	IAM Continuity Indicator: continuity check required on this circuit; COT Continuity Indicator: continuity check successful;	
Comments:	SIP INVITE → 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE →	SUT → ISUP IAM COT

TP101008	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.1 (1,ii,b)
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	PICS 1/6 AND NOT PICS 4/1	
Test purpose:	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, without an SDP offer and reliable provisional responses are supported: <ul style="list-style-type: none"> • the SUT shall immediately send an SDP offer including a media description with both A-law (PCMA) and μ-law (PCMU) included and μ-law (PCMU) shall take precedence over A-law (PCMA) within a 183 Session Progress message; • sending of the IAM shall be deferred until all preconditions have been met. 	
SIP Parameter values:	SIP: 183 SDP1; PRACK SDP2	
ISUP Parameter values:	IAM;	
Comments:	SIP INVITE → 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE → 200 OK UPDATE ← →	SUT ISUP IAM

TP101009	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,1)
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	NOT PICS 4/5	
ISUP selection criteria:	PICS 1/6	
Test purpose:	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer . <ul style="list-style-type: none"> • the SUT shall delete μ-law (PCMU), if present, from the media description that it will send back in the SDP answer; • the SUT shall immediately send out the IAM. 	
SIP Parameter values:	SIP INVITE: Audio RTP/AVP 0, 200 OK: Audio RTP/AVP 8;	
ISUP Parameter values:	IAM USI: A-law or absent;	
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← Conversation	SUT ISUP IAM ACM ANM Conversation Conversation BYE → → REL 200 OK BYE ← ← RLC

TP101010	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,2ai)																																				
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/																																					
SIP selection criteria:	PICS 4/4 AND PICS 4/5																																					
ISUP selection criteria:	PICS 1/4 AND NOT PICS 1/6 AND PICS 4/1																																					
Test purpose:	<p>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer.</p> <ul style="list-style-type: none"> the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer; the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "COT to be expected". 																																					
SIP Parameter values:	SIP INVITE: Audio RTP/AVP 0, 200 OK: Audio RTP/AVP 8;																																					
ISUP Parameter values:	IAM Continuity Indicator: COT to be expected, USI: A-law or absent; COT; Continuity Indicator: continuity;																																					
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 20%;">SIP</td> <td style="width: 20%; text-align: center;">SUT</td> <td style="width: 20%; text-align: center;">ISUP</td> </tr> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">IAM</td> </tr> <tr> <td>183 Session Progress</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>PRACK</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>200 OK PRACK</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>UPDATE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">COT</td> </tr> <tr> <td>200 OK UPDATE</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: center;">ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">ANM</td> </tr> <tr> <td></td> <td style="text-align: center;">Conversation</td> <td style="text-align: center;">Conversation</td> </tr> <tr> <td>BYE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">REL</td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">RLC</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	IAM	183 Session Progress	←		PRACK	→		200 OK PRACK	←		UPDATE	→	COT	200 OK UPDATE	←		180 Ringing	←	ACM	200 OK INVITE	←	ANM		Conversation	Conversation	BYE	→	REL	200 OK BYE	←	RLC	
SIP	SUT	ISUP																																				
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200 OK INVITE	←	ANM																																				
	Conversation	Conversation																																				
BYE	→	REL																																				
200 OK BYE	←	RLC																																				

TP101011	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,2aii)																																				
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/																																					
SIP selection criteria:	PICS 4/4 AND PICS 4/5																																					
ISUP selection criteria:	PICS 1/5 AND NOT PICS 1/6 AND PICS 4/1																																					
Test purpose:	<p>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer.</p> <ul style="list-style-type: none"> the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer; the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "<i>continuity check required on this circuit</i>". 																																					
SIP Parameter values:	SIP INVITE: Audio RTP/AVP 0, 200 OK: Audio RTP/AVP 8;																																					
ISUP Parameter values:	IAM Continuity Indicator: continuity check required on this circuit, USI: A-law or absent COT Continuity Indicator: continuity check successful;																																					
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 20%;">SIP</td> <td style="width: 20%; text-align: center;">SUT</td> <td style="width: 20%; text-align: center;">ISUP</td> </tr> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">IAM</td> </tr> <tr> <td>183 Session Progress</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>PRACK</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>200 OK PRACK</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>UPDATE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">COT</td> </tr> <tr> <td>200 OK UPDATE</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: center;">ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">ANM</td> </tr> <tr> <td></td> <td style="text-align: center;">Conversation</td> <td style="text-align: center;">Conversation</td> </tr> <tr> <td>BYE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">REL</td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">RLC</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	IAM	183 Session Progress	←		PRACK	→		200 OK PRACK	←		UPDATE	→	COT	200 OK UPDATE	←		180 Ringing	←	ACM	200 OK INVITE	←	ANM		Conversation	Conversation	BYE	→	REL	200 OK BYE	←	RLC	
SIP	SUT	ISUP																																				
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UPDATE	→	COT																																				
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200 OK INVITE	←	ANM																																				
	Conversation	Conversation																																				
BYE	→	REL																																				
200 OK BYE	←	RLC																																				

TP101012	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,2b)																																				
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/																																					
SIP selection criteria:	PICS 4/4 AND PICS 4/5																																					
ISUP selection criteria:	NOT PICS 1/6 AND PICS 4/1																																					
Test purpose:	<p>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer.</p> <ul style="list-style-type: none"> the SUT shall delete µ-law (PCMU), if present, from the media description that it will send back in the SDP answer; the shall be deferred until all preconditions have been met. 																																					
SIP Parameter values:	SIP INVITE: Audio RTP/AVP 0, 200 OK: Audio RTP/AVP 8																																					
ISUP Parameter values:	IAM USI: A-law or absent																																					
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">SIP</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td></td> </tr> <tr> <td>183 session Progress</td> <td>←</td> <td></td> </tr> <tr> <td>PRACK</td> <td>→</td> <td></td> </tr> <tr> <td>200 OK PRACK</td> <td>←</td> <td></td> </tr> <tr> <td>UPDATE</td> <td>→</td> <td>→ IAM</td> </tr> <tr> <td>200 OK UPDATE</td> <td>←</td> <td></td> </tr> <tr> <td>180 Ringing</td> <td>←</td> <td>← ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td>←</td> <td>← ANM</td> </tr> <tr> <td></td> <td></td> <td>Conversation</td> </tr> <tr> <td>BYE</td> <td>→</td> <td>→ REL</td> </tr> <tr> <td>200 OK BYE</td> <td>←</td> <td>← RLC</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→		183 session Progress	←		PRACK	→		200 OK PRACK	←		UPDATE	→	→ IAM	200 OK UPDATE	←		180 Ringing	←	← ACM	200 OK INVITE	←	← ANM			Conversation	BYE	→	→ REL	200 OK BYE	←	← RLC	
SIP	SUT	ISUP																																				
INVITE	→																																					
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		Conversation																																				
BYE	→	→ REL																																				
200 OK BYE	←	← RLC																																				

TP101013	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,1)																					
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/																						
SIP selection criteria:	NOT 4/5																						
ISUP selection criteria:	PICS 1/6																						
Test purpose:	<p>Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer:</p> <ul style="list-style-type: none"> the SUT shall delete A-law (PCMA) if both A-law (PCMA) and µ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer; the SUT shall immediately send out the IAM. 																						
SIP Parameter values:	SIP INVITE: Audio RTP/AVP 0 8, 200 OK: Audio RTP/AVP 0; complete called party information																						
ISUP Parameter values:	IAM USI: µ-law																						
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">SIP</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td>→ IAM</td> </tr> <tr> <td>180 Ringing</td> <td>←</td> <td>← ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td>←</td> <td>← ANM</td> </tr> <tr> <td></td> <td></td> <td>Conversation</td> </tr> <tr> <td>BYE</td> <td>→</td> <td>→ REL</td> </tr> <tr> <td>200 OK BYE</td> <td>←</td> <td>← RLC</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	→ IAM	180 Ringing	←	← ACM	200 OK INVITE	←	← ANM			Conversation	BYE	→	→ REL	200 OK BYE	←	← RLC	
SIP	SUT	ISUP																					
INVITE	→	→ IAM																					
180 Ringing	←	← ACM																					
200 OK INVITE	←	← ANM																					
		Conversation																					
BYE	→	→ REL																					
200 OK BYE	←	← RLC																					

TP101014	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,2ai)
TSS reference:		
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	PICS 1/4 AND PICS 1/6 AND PICS 4/1	
Test purpose:	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer :	<ul style="list-style-type: none"> • the SUT shall delete A-law (PCMA) if both A-law (PCMA) and μ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer; • the IAM shall be sent out immediately on the BICC side with the coding of the Nature of Connection Indicators parameter: "COT to be expected".
SIP Parameter values:	SIP INVITE: Audio RTP/AVP 0 8, 200 OK: Audio RTP/AVP 0	
ISUP Parameter values:	IAM USI: μ-law	
Comments:	SIP INVITE → 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE → 200 OK UPDATE ← 180 Ringing ← 200 OK INVITE ← BYE → 200 OK BYE ←	SUT → ISUP IAM COT ACM ANM Conversation REL RLC

TP101015	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,2ai)																																				
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/																																					
SIP selection criteria:	PICS 4/4 AND PICS 4/5																																					
ISUP selection criteria:	PICS 1/5 AND PICS 1/6 AND PICS 4/1																																					
Test purpose:	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer : <ul style="list-style-type: none"> the SUT shall delete A-law (PCMA) if both A-law (PCMA) and μ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer; the IAM shall be sent out immediately on the ISUP side with the Continuity check indicator "<i>continuity check required on this circuit</i>". 																																					
SIP Parameter values:	SIP INVITE: Audio RTP/AVP 0 8, 200 OK: Audio RTP/AVP 0																																					
ISUP Parameter values:	IAM USI: μ-law																																					
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">SIP</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">ISUP</td> </tr> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">IAM</td> </tr> <tr> <td>183 Session Progress</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>PRACK</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>200 OK PRACK</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>UPDATE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">COT</td> </tr> <tr> <td>200 OK UPDATE</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: center;">ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">ANM</td> </tr> <tr> <td></td> <td style="text-align: center;">Conversation</td> <td style="text-align: center;">Conversation</td> </tr> <tr> <td>BYE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">REL</td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">RLC</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	IAM	183 Session Progress	←		PRACK	→		200 OK PRACK	←		UPDATE	→	COT	200 OK UPDATE	←		180 Ringing	←	ACM	200 OK INVITE	←	ANM		Conversation	Conversation	BYE	→	REL	200 OK BYE	←	RLC	
SIP	SUT	ISUP																																				
INVITE	→	IAM																																				
183 Session Progress	←																																					
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180 Ringing	←	ACM																																				
200 OK INVITE	←	ANM																																				
	Conversation	Conversation																																				
BYE	→	REL																																				
200 OK BYE	←	RLC																																				

TP101016	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,2b)																																	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/																																		
SIP selection criteria:	PICS 4/4 AND PICS 4/5																																		
ISUP selection criteria:	PICS 1/6 AND PICS 4/1																																		
Test purpose:	Ensure that if the SUT upon receipt of the first INVITE with sufficient digits, with an SDP offer . <ul style="list-style-type: none"> the SUT shall delete A-law (PCMA) if both A-law (PCMA) and μ-law (PCMU) were present in the offer of the media description, that it will send it back in the SDP answer; the IAM shall be deferred until all preconditions have been met. 																																		
SIP Parameter values:	SIP INVITE: Audio RTP/AVP 0 8, 200 OK: Audio RTP/AVP 0;																																		
ISUP Parameter values:	IAM USI: μ-law;																																		
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">SIP</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">ISUP</td> </tr> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>PRACK</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>200 OK PRACK</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>UPDATE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">IAM</td> </tr> <tr> <td>200 OK UPDATE</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: center;">ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">ANM</td> </tr> <tr> <td></td> <td style="text-align: center;">Conversation</td> <td style="text-align: center;">Conversation</td> </tr> <tr> <td>BYE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">REL</td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">RLC</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→		PRACK	→		200 OK PRACK	←		UPDATE	→	IAM	200 OK UPDATE	←		180 Ringing	←	ACM	200 OK INVITE	←	ANM		Conversation	Conversation	BYE	→	REL	200 OK BYE	←	RLC	
SIP	SUT	ISUP																																	
INVITE	→																																		
PRACK	→																																		
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UPDATE	→	IAM																																	
200 OK UPDATE	←																																		
180 Ringing	←	ACM																																	
200 OK INVITE	←	ANM																																	
	Conversation	Conversation																																	
BYE	→	REL																																	
200 OK BYE	←	RLC																																	

TP101017	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1] clauses 6.1.3.1, 6.1.3.2, 6.1.3.3 and 6.1.3.4																					
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/																						
SIP selection criteria:	PICS 1/2																						
ISUP selection criteria:	NOT PICS 1/9																						
Test purpose:	Ensure that the SUT on receipt of an INVITE message: <ul style="list-style-type: none"> • sends an IAM message, where the Calling party's category is set to "Ordinary calling subscriber", the Nature of Connection Indicators (NCI) encoded as follows: • Satellite indicator set to: "One satellite circuit in the connection" Continuity check indicator set to: one satellite in the connection 																						
SIP Parameter values:																							
ISUP Parameter values:	Nature of Connection Indicators (NCI): Satellite indicator set to: "One satellite circuit in the connection"																						
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP</th> <th style="text-align: center; width: 40%;">SUT</th> <th style="text-align: right; width: 30%;">ISUP</th> </tr> </thead> <tbody> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">IAM</td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: right;">ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">ANM</td> </tr> <tr> <td></td> <td style="text-align: center;">Conversation</td> <td style="text-align: right;">Conversation</td> </tr> <tr> <td>BYE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">REL</td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">RLC</td> </tr> </tbody> </table>	SIP	SUT	ISUP	INVITE	→	IAM	180 Ringing	←	ACM	200 OK INVITE	←	ANM		Conversation	Conversation	BYE	→	REL	200 OK BYE	←	RLC	
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BYE	→	REL																					
200 OK BYE	←	RLC																					

TP101018	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.1.3.1, 6.1.3.2, 6.1.3.3 and 6.1.3.4	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/		
SIP selection criteria:	PICS 1/2		
ISUP selection criteria:	PICS 1/9 AND NOT PICS 4/23		
Test purpose:	Ensure that the SUT on receipt of an INVITE message: <ul style="list-style-type: none"> • sends an IAM message, where the Calling party's category is set to "Ordinary calling subscriber", the Nature of Connection Indicators (NCI) encoded as follows: <ul style="list-style-type: none"> - Satellite indicator set to: "One satellite circuit in the connection". - the Forward call indicator is encoded as follows <ul style="list-style-type: none"> Interworking indicator: Interworking encountered ISUP/BICC Indicator: ISDN User part/BICC not used all the way ISUP/BICC Preference indicator: ISDN user part/BICC not required all the way ISDN access indicator: Originating access non-ISDN. 		
SIP Parameter values:			
ISUP Parameter values:	Nature of Connection Indicators (NCI): Satellite indicator set to: "One satellite circuit in the connection" Forward Call Indicators (FCI): Interworking indicator: interworking encountered ISDN user part indicator: ISDN user part/BICC not used all the way ISDN access indicator: originating access non-ISDN ISDN user part preference indicator: ISDN user part/BICC not required all the way		
Comments:	SIP INVITE 180 Ringing 200 OK INVITE BYE 200 OK BYE	SUT Conversation Conversation	ISUP IAM ACM ANM REL RLC

TP101019	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.1.3.1, 6.1.3.2, 6.1.3.3 and 6.1.3.4																					
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/																						
SIP selection criteria:	PICS 1/1 AND NOT PICS 4/24																						
ISUP selection criteria:																							
Test purpose:	<p>Ensure that the SUT on receipt of an INVITE message:</p> <ul style="list-style-type: none"> • sends an IAM message, where the Calling party's category is set to "Ordinary calling subscriber", the Nature of Connection Indicators (NCI) encoded as follows: <ul style="list-style-type: none"> - Satellite indicator set to: one satellite circuit in the connection" - Echo control device indicator set to: "Outgoing echo control device included". - the Forward call indicator is encoded as follows <ul style="list-style-type: none"> Interworking indicator: Interworking encountered ISUP/BICC Indicator: ISDN User part/BICC not used all the way ISUP/BICC Preference indicator: ISDN user part/BICC not required all the way ISDN access indicator: Originating access non-ISDN 																						
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TP101020	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.3.5																					
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/																						
SIP selection criteria:	Based on table 1																						
ISUP selection criteria:																							
Test purpose:	Ensure that the SUT in the Idle state on receipt of an INVITE message containing the media description defined with the "a = " "b =" and "m=" lines set to a_b_m_LINE_VALUE: <ul style="list-style-type: none"> • sends an IAM message, with the Transmission Medium Requirement (TMR) parameter set to TMR_VALUE. 																						
SIP Parameter values:	INVITE; a_b_m_LINE_VALUE																						
ISUP Parameter values:	IAM; TMR: ISUP_TMR																						
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 33%;">SIP</td> <td style="width: 33%; text-align: center;">SUT</td> <td style="width: 33%;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>➔</td> <td>IAM</td> </tr> <tr> <td>180 Ringing</td> <td>⬅</td> <td>ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td>⬅</td> <td>ANM</td> </tr> <tr> <td></td> <td>Conversation</td> <td>Conversation</td> </tr> <tr> <td>BYE</td> <td>➔</td> <td>REL</td> </tr> <tr> <td>200 OK BYE</td> <td>⬅</td> <td>RLC</td> </tr> </table>	SIP	SUT	ISUP	INVITE	➔	IAM	180 Ringing	⬅	ACM	200 OK INVITE	⬅	ANM		Conversation	Conversation	BYE	➔	REL	200 OK BYE	⬅	RLC	
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200 OK BYE	⬅	RLC																					

Table 1

Values for test purposes TP101020						
	a_b_m_LINE_VALUE					
	m= line		b= line	a= line	TMR_VALUE	
test purposes	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT><encoding name>/<clock rate>[/encoding parameters>	TMR codes
				see note 1		
VA_01	audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	"3,1KHz audio"
VA_02	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT>PCMU/8000	"3,1KHz audio"
VA_03	audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	"3,1KHz audio"
VA_04	audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT>PCMA/8000	"3,1KHz audio"
VA_05	audio	RTP/AVP	9	AS:64 kbit/s	rtpmap:9 G722/8000	"64 kbit/s preferred"
VA_06	audio	RTP/AVP	Dynamic PT	AS:64 kbit/s	rtpmap:<dynamic-PT>CLEARMODE/8000 (see note 2)	"64 kbit/s unrestricted"
VA_07	image	Udptl	t38	N/A or up to 64 kbit/s	Based on T.38	"3,1 KHz audio"
VA_08	image	Tcptl	t38	N/A or up to 64 kbit/s	Based on T.38	"3,1 KHz audio"

NOTE 1: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.

NOTE 2: CLEARMODE has been standardized.

TP101021	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.3.5																					
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/																						
SIP selection criteria:	Based on table 2																						
ISUP selection criteria:																							
Test purpose:	Ensure that the SUT in the Idle state on receipt of an INVITE message, with the media description defined with the "a = " "b =" and "m=" lines set to a_b_m_LINE_VALUE: <ul style="list-style-type: none"> • sends an IAM message, with the user information parameter set to USI_VALUE 																						
SIP Parameter values:	INVITE: a_b_m_LINE_VALUE																						
ISUP Parameter values:																							
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP</th> <th style="text-align: center; width: 40%;">SUT</th> <th style="text-align: right; width: 30%;">ISUP</th> </tr> </thead> <tbody> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">IAM</td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: right;">ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">ANM</td> </tr> <tr> <td></td> <td style="text-align: center;">Conversation</td> <td style="text-align: right;">Conversation</td> </tr> <tr> <td>BYE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">REL</td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">RLC</td> </tr> </tbody> </table>	SIP	SUT	ISUP	INVITE	→	IAM	180 Ringing	←	ACM	200 OK INVITE	←	ANM		Conversation	Conversation	BYE	→	REL	200 OK BYE	←	RLC	
SIP	SUT	ISUP																					
INVITE	→	IAM																					
180 Ringing	←	ACM																					
200 OK INVITE	←	ANM																					
	Conversation	Conversation																					
BYE	→	REL																					
200 OK BYE	←	RLC																					

Table 2

Values for test purposes TP101021							
	a_b_m_LINE_VALUE						
	m= line			b= line	A= line	USI_VALUE	
test purposes	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT><encoding name>/<clock rate>[/<encoding parameters>]	Information Transport Capability	User Information Layer 1 Protocol Indicator
				see note 1			
VA_01	Audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	"3,1KHz audio"	"G.711 μ-law"
VA_02	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT>PCMU/8000	"3,1KHz audio"	"G.711 μ-law"
VA_03	Audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	"3,1KHz audio"	"G.711 A-law"
VA_04	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT>PCMA/8000	"3,1KHz audio"	"G.711 A-law"
VA_05	Audio	RTP/AVP	9	AS:64 kbit/s	rtpmap:9 G722/8000	"Unrestricted digital inf. w/tones/ann"	
VA_06	Audio	RTP/AVP	Dynamic PT	AS:64 kbit/s	rtpmap:<dynamic-PT>CLEARMODE/8000 (see note 2)	"Unrestricted digital information"	
VA_07	image	Udptl	t38	N/A or up to 64 kbit/s	Based on T.38	"3,1KHz audio"	
VA_08	image	Tcptl	t38	N/A or up to 64 kbit/s	Based on T.38	"3,1KHz audio"	

NOTE 1: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.

NOTE 2: CLEARMODE has been standardized.

TP101022	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.3.5	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/		
SIP selection criteria:	Based on table 3		
ISUP selection criteria:			
Test purpose:	Ensure that the SUT in the Idle state on receipt of an INVITE message, with the media description defined with the "a =" "b =" and "m=" lines to lines set to a_b_m_LINE_VALUE: <ul style="list-style-type: none"> • sends an IAM message with the Access transport parameter containing the HLC information element. 		
SIP Parameter values:	INVITE: a_b_m_LINE_VALUE		
ISUP Parameter values:	IAM; Access transport parameter HLC : HLC_VALUE		
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← BYE → 200 OK BYE ←	SUT Conversation Conversation → ←	ISUP IAM ACM ANM REL RLC

Table 3

Values for test purposes TP101022						
	M= line			b= line	a= line	HLC parameter HLC_VALUE
Test purposes	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>	HLC_VALUE
				see note 1		
VA_01	Audio	RTP/AVP	0	N/A or up to 64 kbit/s	N/A	See note 2
VA_02	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMU/8000	See note 2
VA_03	Audio	RTP/AVP	8	N/A or up to 64 kbit/s	N/A	See note 2
VA_04	Audio	RTP/AVP	Dynamic PT	N/A or up to 64 kbit/s	rtpmap:<dynamic-PT> PCMA/8000	See note 2
VA_05	Image	Udptl	t38	N/A or up to 64 kbit/s	Based on T.38	"Facsimile Group 2/3"
VA_06	Image	Tcptl	t38	N/A or up to 64 kbit/s	Based on T.38	"Facsimile Group 2/3"

NOTE 1: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.

NOTE 2: HLC normally absent in this case. It is possible for HLC to be present with the value "Telephony", although clause 6.3.1/ITU-T Rec Q.939 [13] indicates that this would normally be accompanied by a value of "Speech" for the Information Transfer Capability element.

TP101023	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.3.9
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:	PICS 4/3	
Test purpose:	Ensure that the SUT for Profiles A and B the I-IWU shall derive the Hop Counter parameter value from the Max-Forwards header field value by applying a factor. The Hop Counter for a given message should never increase and should decrease by at least 1 with each successive visit to an IWU, regardless of intervening interworking, and similarly for Max-Forwards in the SIP domain.	
SIP Parameter values:		
ISUP Parameter values:	IAM: Hop Counter parameter value	
Comments:	SIP INVITE → SUT → ISUP 180 Ringing ← ← IAM 200 OK INVITE ← ← ACM The initial and successively mapped values of Hop Counter should be large enough to accommodate the maximum number of hops that might be expected of a validly routed call.	

TP101024	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,1)
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 1/9	
ISUP selection criteria:	NOT PICS 1/7	
Test purpose:	Ensure that the SUT on receipt of an INVITE message with a Called party number contained in the userinfocomponent of the Request-URI: <ul style="list-style-type: none"> Nature of address indicator: Analyse the information contained in received URI with user=phone, and if it is in the format: +CC NDC SN where CC is the country code of the network in which the next hop terminates, then set Nature of Address indicator to "National (significant) number", remove "+CC" and use the remaining digits to fill the Address signals". Internal Network Number Indicator: routing to internal network number not allowed. <u>Numbering plan Indicator: 001 ISDN (Telephony) numbering plan.</u> Address Signals: NDC SN. 	
SIP Parameter values:		
ISUP Parameter values:	IAM: Called party number	
Comments:	SIP INVITE → SUT → ISUP 180 Ringing ← ← IAM 200 OK INVITE ← ← ACM Conversation ← ← ANM BYE → → REL 200 OK BYE ← ← RLC	

TP101025	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,1)																																				
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/																																					
SIP selection criteria:	PICS 1/9																																					
ISUP selection criteria:	PICS 1/7																																					
Test purpose:	<p>Ensure that the SUT on receipt of an INVITE message with a Called party number contained in the userinfo component of the Request-URI:</p> <ul style="list-style-type: none"> Nature of address indicator: Analyse the information contained in received URI with user=phone, and if it is in the format: +CC NDC SN where CC is not the country code of the network in which the next hop terminates, then set Nature of Address indicator to "International number", remove "+" and use the remaining digits to fill the Address signals. Internal Network Number Indicator: routing to internal network number not allowed. Numbering plan Indicator: 001 ISDN (Telephony) numbering plan. Address Signals CC NDC SN. 																																					
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TP101026	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,1)																																							
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/																																								
SIP selection criteria:	NOT PICS 1/9																																								
ISUP selection criteria:																																									
Test purpose:	<p>Ensure that the SUT on receipt of an INVITE message with a Called party number contained in the userinfo component of the Request-URI:</p> <ul style="list-style-type: none"> Internal Network Number Indicator: routing to internal network number not allowed. Numbering plan Indicator: 001 ISDN (Telephony) numbering plan. Address Signals. 																																								
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	Conversation	Conversation																																							
BYE	→	→																																							
200 OK BYE	←	←																																							
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TP101027	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,1)	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/		
SIP selection criteria:	PICS 1/9		
ISUP selection criteria:	PICS 1/7		
Test purpose:	Ensure that the SUT on receipt of an INVITE message with a SDP offer for μ-Law and a-Law, then independent from the received order of preference : <ul style="list-style-type: none"> • the G.711 a-law codec shall be returned in the SDP answer as preferred codec. 		
SIP Parameter values:	Offer: m=audio 4711 RTP/AVP 0 8 Answer: m=audio 4712 RTP/AVP 8 0		
ISUP Parameter values:			
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← Conversation BYE → 200 OK BYE ←	SUT → ← ← Conversation → ←	ISUP IAM ACM ANM Conversation REL RLC

TP101028	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,1)	
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/		
SIP selection criteria:	PICS 1/9		
ISUP selection criteria:	PICS 1/7		
Test purpose:	Ensure that the SUT on receipt of an INVITE message with a SDP offer for a-Law and no μ-Law, then independent the normal offer answer procedures apply : <ul style="list-style-type: none"> • the G.711 a-law codec shall be returned in the SDP answer. 		
SIP Parameter values:	Offer: m=audio 4711 RTP/AVP 8 Answer: m=audio 4711 RTP/AVP 8		
ISUP Parameter values:			
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← Conversation BYE → 200 OK BYE ←	SUT → ← ← Conversation → ←	ISUP IAM ACM ANM Conversation REL RLC

TP101029	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,1)
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 1/9	
ISUP selection criteria:	PICS 1/7	
Test purpose:	Ensure that the SUT on receipt of an INVITE message with a SDP offer without a-law codec : • the u-law codec shall be rejected.	
SIP Parameter values:	Offer: m=audio 4711 RTP/AVP 0 Answer: m=audio 0 RTP/AVP 0	
ISUP Parameter values:		
Comments:	SIP CASE A INVITE 180 Ringing 200 OK INVITE	SUT → ← ← SUT ISUP IAM ACM ANM

TP101030	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,1)
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 1/9 AND PICS 4/19	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams and based on operator policy then : • the call is refused with a 415 Unsupported media type response.	
SIP Parameter values:	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8	
ISUP Parameter values:		
Comments:	SIP CASE A INVITE 415 Unsupported media type ACK	SUT → ← → SUT ISUP

TP101031	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.1.2 (i,1)
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 1/9 AND NOT PICS 4/19	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT on receipt of an INVITE message with a SDP offer with more than one media streams and based on operator policy then : <ul style="list-style-type: none"> if the SDP offer contains one or more audio type media streams and one or more non-audio type media stream, only the audio streams shall be considered; the other streams shall be rejected; if the SDP offer contains several audio type media streams, the IWU shall only consider one, and reject the other streams. 	
SIP Parameter values:	Offer: m=audio 4711 RTP/AVP 8 m= audio 4712 RTP/AVP 8 m= video 4713 RTP/AVP 31 Answer: m=audio 4711 RTP/AVP 8 m=audio 0 RTP/AVP 8 m=video 0 RTP/AVP 31	
ISUP Parameter values:		
Comments:	SIP CASE A INVITE 180 Ringing 200 OK INVITE	SUT → ← ← → IAM ACM ANM

TP101032	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.1.3.1, 6.1.3.2, 6.1.3.3 and 6.1.3.4
TSS reference:	SIP-ISUP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:	PICS 1/2 AND PICS 1/9 AND PICS 4/23	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT on receipt of an INVITE message: <ul style="list-style-type: none"> sends an IAM message, where the Calling party's category is set to "Ordinary calling subscriber", the Nature of Connection Indicators (NCI) encoded as follows: <ul style="list-style-type: none"> Satellite indicator set to: "One satellite circuit in the connection". the Forward call indicator is encoded as follows <ul style="list-style-type: none"> Interworking indicator: No interworking encountered ISUP/BICC Indicator: ISDN User part/BICC used all the way ISUP/BICC Preference indicator: ISDN user part/BICC not required all the way ISDN access indicator: Originating access ISDN. 	
SIP Parameter values:		
ISUP Parameter values:	Nature of Connection Indicators (NCI): Satellite indicator set to: "One satellite circuit in the connection" Forward Call Indicators (FCI): Interworking indicator: No interworking encountered ISDN user part indicator: ISDN user part/BICC used all the way ISDN access indicator: originating access ISDN ISDN user part preference indicator: ISDN user part/BICC not required all the way	

6.2.1.2 Sending of the Subsequent Address Message (SAM)

TP102001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.2 a)			
TSS reference:	SIP-ISUP/Basic call/ Sending of the Subsequent Address Message (SAM)/				
SIP selection criteria:	PICS 3/4				
ISUP selection criteria:	PICS 3/8				
Test purpose:	<p>Ensure that the SUT receives an INVITE with the same Call-ID and From tag as a previous INVITE which was associated with a BICC/ISUP call/bearer control instance currently existing on the BICC/ISUP side whereby the number of digits in the Request-URI is greater than the number of digits already accumulated for the call:</p> <ul style="list-style-type: none"> • Sends a SAM and pass it to outgoing BICC/ISUP procedures. • The SAM shall contain in its Subsequent Number parameter only the additional digits received in this Request-URI compared with the digits already accumulated for the call. 				
SIP Parameter values:					
ISUP Parameter values:	SAM; subsequent number (PIXIT)				
Comments:	SIP INVITE INVITE INVITE 180 Ringing 200 OK INVITE	SUT → → → ← ←	ISUP IAM SAM SAM ACM ANM		

TP102002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.2 b)			
TSS reference:	SIP-ISUP/Basic call/ Sending of the Subsequent Address Message (SAM)/				
SIP selection criteria:	PICS 3/4				
ISUP selection criteria:	PICS 3/8				
Test purpose:	<p>Ensure that the SUT receives an INVITE with the same Call-ID and From tag as a previous INVITE which was associated with a BICC/ISUP call/bearer control instance currently existing on the BICC/ISUP side whereby the number of digits in the Request-URI is fewer than the number of digits already accumulated for the call:</p> <ul style="list-style-type: none"> • Then the SUT shall immediately send a 484 Address Incomplete response for this INVITE. • In this case no SAM is sent to BICC/ISUP procedures. 				
SIP Parameter values:					
ISUP Parameter values:					
Comments:	SIP INVITE INVITE 484 Address incomplete	SUT → → ←	ISUP IAM		

6.2.1.3 Sending of COT

TP103001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.3
TSS reference:	SIP-ISUP/Basic call/COT	
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	PICS 1/4 AND PICS 4/1	
Test purpose:	<p>Ensure that the when the SUT determines that all the preconditions on the incoming SIP side have been met and any continuity procedures on the outgoing BICC side have been successfully completed:</p> <ul style="list-style-type: none"> the SUT shall send the COT message where the Continuity Indicator in the COT message shall be set to "Continuity". 	
SIP Parameter values:		
ISUP Parameter values:	COT continuity indicator: Continuity;	
Comments:	SIP INVITE → 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE → 200 OK UPDATE ← 180 Ringing ← 200 OK INVITE ←	SUT → BICC IAM COT ACM ANM

TP103002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.3
TSS reference:	SIP-ISUP/Basic call/ COT	
SIP selection criteria:	PICS 4/4 AND PICS 4/5	
ISUP selection criteria:	PICS 1/5 AND PICS 4/1	
Test purpose:	<p>Ensure that the when the SUT determines that all the preconditions on the incoming SIP side have been met and any continuity procedures on the outgoing ISUP side have been successfully completed:</p> <ul style="list-style-type: none"> the I-IWU shall send the COT message where the Continuity Indicator in the COT message shall be set to "<i>Continuity check successful</i>". 	
SIP Parameter values:		
ISUP Parameter values:	COT continuity indicator: Continuity check successful;	
Comments:	SIP INVITE → 183 Session Progress ← PRACK → 200 OK PRACK ← UPDATE → 200 OK UPDATE ← 200 OK INVITE ←	SUT → → ISUP IAM COT ANM

6.2.1.4 Receipt of the Address Complete Message (ACM)

TP104001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.5, 2)
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT on receipt of an ACM message where the Called party status indicator is set to "no indication": <ul style="list-style-type: none">• in the case of Profile A or Profile B, the ACM is not interworked.	
SIP Parameter values:		
ISUP Parameter values:	ACM Called party status: no indication;	
Comments:	SIP INVITE	SUT → ISUP IAM ACM ← Ringing tone

TP104002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.5, 1)
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Address complete message (ACM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT on receipt of an ACM message where the Called party status indicator is set to "subscriber free" where the ISUP indicator parameter set to ISUP_ID, the ISDN access indicator set to ISDN_ACCESES_ID and the OBCI in-band information set to OBCI_INBAND then: <ul style="list-style-type: none">• in case of Profile A or Profile B, the 180 Ringing SIP response is sent. Ensure that the in-band information can be transmitted to the calling user.	
SIP Parameter values:		
ISUP Parameter values:	ACM FCI: ISUP_ID, ISDN_ACCESS_ID OBCI: OBCI_INBAND;	
Comments:	SIP INVITE 180 Ringing	SUT → Ringing tone ← Ringing tone
		ISUP IAM ACM ← Ringing tone

Table 4

Test purposes	ISUP Parameter values:
VA_01	ACM ISUP_ID: ISUP not used all the way OBCI_INBAND: no
VA_02	ACM ISUP_ID: ISUP not used all the way OBCI_INBAND: yes
VA_03	ACM ISUP_ID: ISUP used all the way ISDN_ACSES_ID: non ISDN OBCI_INBAND: no
VA_04	ACM ISUP_ID: ISUP used all the way ISDN_ACSES_ID: non ISDN OBCI_INBAND: yes
VA_05	ACM ISUP_ID: ISUP used all the way ISDN access indicator: ISDN OBCI_INBAND: yes

6.2.1.5 Receipt of the Call progress message (CPG)

TP105001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.6
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT, having received the ACM message called party status indicator "no indication", on receipt of a CPG message where the event information parameter event indicator is set to "Alerting": • the 180 Ringing SIP response is sent.	
SIP Parameter values:		
ISUP Parameter values:	ACM: Called party status "no indication" CPG; event information parameter event indicator : Alerting	
Comments:	SIP INVITE 180 Ringing	SUT → IAM ← ACM ← CPG

TP105002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.6
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT, having received the ACM message called party status indicator "no indication", on receipt of a CPG message where the event information parameter event indicator is set to "Progress": <ul style="list-style-type: none">• the CPG is not interworked.	
SIP Parameter values:		
ISUP Parameter values:	ACM: Called party status "no indication" CPG; event information parameter event indicator : Progress	
Comments:	SIP INVITE	SUT → ← ← ISUP IAM ACM CPG

TP105003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.6
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Call progress message (CPG).	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT, having received the ACM message called party status indicator "no indication", on receipt of a CPG message where the event information parameter event indicator is set to " <i>in-band information or an appropriate pattern is now available</i> ": <ul style="list-style-type: none">• the CPG is not interworked.	
SIP Parameter values:		
ISUP Parameter values:	ACM: Called party status "no indication" CPG; event information parameter event indicator : in-band-information or an appropriate pattern is now available	
Comments:	SIP INVITE	SUT → ← ← ISUP IAM ACM CPG

6.2.1.6 Receipt of the Answer message (ANM)

TP106001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.7												
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).													
SIP selection criteria:														
ISUP selection criteria:														
Test purpose:	Ensure that the SUT, having received the ACM message Called party status indicator set to "subscriber free", on receipt of an ANM message: <ul style="list-style-type: none"> • sends a 200 OK INVITE to the UAC. 													
SIP Parameter values:														
ISUP Parameter values:														
Comments:	<p>The bearer path shall be connected in both directions when both of the following conditions are satisfied:</p> <ul style="list-style-type: none"> • the BICC outgoing bearer set-up procedure, (ITU-T Rec Q.1902.4 [4]) is successfully completed, and; • the I-IWU determines (using the procedures defined in RFC 3312 [7]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable). <p>In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "<i>notification not required</i>", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [7]) that sufficient preconditions have been met for the session to proceed.</p> <table style="width: 100%; text-align: center;"> <tr> <td style="width: 33%;">SIP</td> <td style="width: 33%;">SUT</td> <td style="width: 33%;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td>IAM</td> </tr> <tr> <td>180 Ringing</td> <td>←</td> <td>ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td>←</td> <td>ANM</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	IAM	180 Ringing	←	ACM	200 OK INVITE	←	ANM	
SIP	SUT	ISUP												
INVITE	→	IAM												
180 Ringing	←	ACM												
200 OK INVITE	←	ANM												

TP106002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.7																					
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).																						
SIP selection criteria:	PICS 4/5																						
ISUP selection criteria:																							
Test purpose:	No SDP offer received in the INVITE. Ensure that the SUT, having received the ACM message Called party status indicator set to "subscriber free", on receipt of an ANM message: <ul style="list-style-type: none"> • sends a 200 OK INVITE to the UAC. 																						
SIP Parameter values:																							
ISUP Parameter values:																							
Comments:	<p>The bearer path shall be connected in both directions when both of the following conditions are satisfied:</p> <ul style="list-style-type: none"> • the BICC outgoing bearer set-up procedure, (ITU-T Rec Q.1902.4 [4]) is successfully completed, and; • the I-IWU determines (using the procedures defined in RFC 3312 [7]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable). <p>In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "<i>notification not required</i>", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [7]) that sufficient preconditions have been met for the session to proceed.</p> <table style="width: 100%; text-align: center;"> <tr> <td style="width: 33%;">SIP</td> <td style="width: 33%;">SUT</td> <td style="width: 33%;">ISUP</td> </tr> <tr> <td>INVITE →</td> <td></td> <td></td> </tr> <tr> <td>183 Session Progress(SDP) ←</td> <td></td> <td></td> </tr> <tr> <td>PRACK(SDP) →</td> <td>→</td> <td>IAM</td> </tr> <tr> <td>200 OK (PRACK) ←</td> <td></td> <td></td> </tr> <tr> <td>180 Ringing ←</td> <td>←</td> <td>ACM</td> </tr> <tr> <td>200 OK INVITE ←</td> <td>←</td> <td>ANM</td> </tr> </table>	SIP	SUT	ISUP	INVITE →			183 Session Progress(SDP) ←			PRACK(SDP) →	→	IAM	200 OK (PRACK) ←			180 Ringing ←	←	ACM	200 OK INVITE ←	←	ANM	
SIP	SUT	ISUP																					
INVITE →																							
183 Session Progress(SDP) ←																							
PRACK(SDP) →	→	IAM																					
200 OK (PRACK) ←																							
180 Ringing ←	←	ACM																					
200 OK INVITE ←	←	ANM																					

TP106003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.7															
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (ANM).																
SIP selection criteria:	NOT PICS 4/5																
ISUP selection criteria:																	
Test purpose:	SDP offer was not received in the initial INVITE. Ensure that the SUT, having received the ACM message, on receipt of an ANM message: <ul style="list-style-type: none"> • sends a 200 OK INVITE to the UAC. The 200 OK INVITE shall include an SDP offer consistent with the TMR/USI used on the BICC/ISUP side. 																
SIP Parameter values:	200 OK INVITE includes an SDP offer ACK includes an SDP answer																
ISUP Parameter values:																	
Comments:	<p>The bearer path shall be connected in both directions when both of the following conditions are satisfied:</p> <ul style="list-style-type: none"> • the BICC outgoing bearer set-up procedure, (ITU-T Rec Q.1902.4 [4]) is successfully completed, and; • the I-IWU determines (using the procedures defined in RFC 3312 [7]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable). <p>In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "<i>notification not required</i>", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [7]) that sufficient preconditions have been met for the session to proceed.</p> <table style="width: 100%; text-align: center;"> <tr> <td style="width: 33.33%;">SIP</td> <td style="width: 33.33%;">SUT</td> <td style="width: 33.33%;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td>IAM</td> </tr> <tr> <td>180 Ringing</td> <td>←</td> <td>ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td>←</td> <td>ANM</td> </tr> <tr> <td>ACK</td> <td>→</td> <td></td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	IAM	180 Ringing	←	ACM	200 OK INVITE	←	ANM	ACK	→		
SIP	SUT	ISUP															
INVITE	→	IAM															
180 Ringing	←	ACM															
200 OK INVITE	←	ANM															
ACK	→																

6.2.1.7 Receipt of the Connect message (CON)

TP107001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.4 and 6.7									
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CONNECT message (CON).										
SIP selection criteria:											
ISUP selection criteria:											
Test purpose:	<p>SDP offer was received in the initial INVITE. Ensure that the SUT, on receipt of an CON message:</p> <ul style="list-style-type: none"> • sends a 200 OK INVITE to the UAC. 										
SIP Parameter values:											
ISUP Parameter values:											
Comments:	<p>The bearer path shall be connected in both directions when both of the following conditions are satisfied:</p> <ul style="list-style-type: none"> • the BICC outgoing bearer set-up procedure, (ITU-T Rec Q.1902.4 [4]) is successfully completed, and; • the I-IWU determines (using the procedures defined in RFC 3312 [7]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable). <p>In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "<i>notification not required</i>", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [7]) that sufficient preconditions have been met for the session to proceed.</p>	<table style="width: 100%; text-align: center;"> <tr> <td style="width: 33%;">SIP</td> <td style="width: 33%;">SUT</td> <td style="width: 33%;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td>IAM</td> </tr> <tr> <td>200 OK INVITE</td> <td>←</td> <td>CON</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	IAM	200 OK INVITE	←	CON
SIP	SUT	ISUP									
INVITE	→	IAM									
200 OK INVITE	←	CON									

TP107002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.4 and 6.7																		
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CONNECT message (CON).																			
SIP selection criteria:	PICS 4/5																			
ISUP selection criteria:																				
Test purpose:	Ensure that the SUT, on receipt of an CON message: <ul style="list-style-type: none"> • sends a 200 OK INVITE to the UAC. 																			
SIP Parameter values:																				
ISUP Parameter values:																				
Comments:	<p>The bearer path shall be connected in both directions when both of the following conditions are satisfied:</p> <ul style="list-style-type: none"> • the BICC outgoing bearer set-up procedure, (ITU-T Rec Q.1902.4 [4]) is successfully completed; and • the I-IWU determines (using the procedures defined in RFC 3312 [7]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable). <p>In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "<i>notification not required</i>", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [7]) that sufficient preconditions have been met for the session to proceed.</p> <table style="width: 100%; text-align: center;"> <tr> <td style="width: 33%;">SIP</td> <td style="width: 33%;">SUT</td> <td style="width: 33%;">ISUP</td> </tr> <tr> <td>INVITE →</td> <td></td> <td></td> </tr> <tr> <td>183 Session Progress(SDP) ←</td> <td></td> <td></td> </tr> <tr> <td>PRACK(SDP) →</td> <td>→</td> <td>IAM</td> </tr> <tr> <td>200 OK PRACK ←</td> <td></td> <td></td> </tr> <tr> <td>200 OK INVITE ←</td> <td>←</td> <td>CON</td> </tr> </table>		SIP	SUT	ISUP	INVITE →			183 Session Progress(SDP) ←			PRACK(SDP) →	→	IAM	200 OK PRACK ←			200 OK INVITE ←	←	CON
SIP	SUT	ISUP																		
INVITE →																				
183 Session Progress(SDP) ←																				
PRACK(SDP) →	→	IAM																		
200 OK PRACK ←																				
200 OK INVITE ←	←	CON																		

TP107003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.4 and 6.7									
TSS reference:	SIP-ISUP/Basic call/ Receipt of the Answer message (CON).										
SIP selection criteria:	NOT PICS 4/5										
ISUP selection criteria:											
Test purpose:	<p>SDP offer was not received in the initial INVITE. Ensure that the SUT, on receipt of an CON message:</p> <ul style="list-style-type: none"> sends a 200 OK INVITE to the UAC. The 200 OK INVITE shall include an SDP offer consistent with the TMR/USI used on the BICC/ISUP side. 										
SIP Parameter values:	200 OK INVITE includes an SDP offer										
ISUP Parameter values:											
Comments:	<p>The bearer path shall be connected in both directions when both of the following conditions are satisfied:</p> <ul style="list-style-type: none"> The BICC outgoing bearer set-up procedure, (ITU-T Rec Q.1902.4 [4]) is successfully completed, and The I-IWU determines (using the procedures defined in RFC 3312 [7]) that sufficient preconditions have been satisfied on the SIP side for session establishment to proceed (if applicable). <p>In addition, if BICC is performing the "Per-call bearer set-up in the forward direction" Outgoing bearer set-up procedure and the Connect Type is "<i>notification not required</i>", the bearer path shall be connected in both directions when the Bearer Set-up request is sent and the I-IWU determines (through the procedures defined in RFC 3312 [7]) that sufficient preconditions have been met for the session to proceed.</p> <table style="width: 100%; text-align: center;"> <tr> <td style="width: 33%;">SIP INVITE 200 OK INVITE</td> <td style="width: 33%;">SUT</td> <td style="width: 33%;">ISUP IAM CON</td> </tr> <tr> <td>→</td> <td>→</td> <td>→</td> </tr> <tr> <td>←</td> <td>←</td> <td>←</td> </tr> </table>	SIP INVITE 200 OK INVITE	SUT	ISUP IAM CON	→	→	→	←	←	←	
SIP INVITE 200 OK INVITE	SUT	ISUP IAM CON									
→	→	→									
←	←	←									

6.2.1.8 Receipt of the REL message

TP108001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.2												
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/													
SIP selection criteria:	NOT PICS 4/10													
ISUP selection criteria:														
Test purpose:	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, on receipt of an ISUP REL:	<ul style="list-style-type: none"> The SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side. The SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA. 												
SIP Parameter values:	SIP Status-Code: SIP_FAILURE_VA (PIXIT)													
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)													
Comments:	<table style="width: 100%; text-align: center;"> <tr> <td style="width: 33%;">SIP INVITE SIP_FAILURE_VA ACK</td> <td style="width: 33%;">SUT</td> <td style="width: 33%;">ISUP IAM REL RLC</td> </tr> <tr> <td>→</td> <td>→</td> <td>→</td> </tr> <tr> <td>←</td> <td>←</td> <td>←</td> </tr> <tr> <td>→</td> <td>→</td> <td>→</td> </tr> </table>	SIP INVITE SIP_FAILURE_VA ACK	SUT	ISUP IAM REL RLC	→	→	→	←	←	←	→	→	→	
SIP INVITE SIP_FAILURE_VA ACK	SUT	ISUP IAM REL RLC												
→	→	→												
←	←	←												
→	→	→												

Table 5

Values for test purpose TP108001	
← SIP Message SIP_FAILURE_VA	← REL Cause Indicators parameter CV_ISUP
VA_1	404 Not Found
VA_2	500 Server Internal Error
VA_3	500 Server Internal Error
VA_4	500 Server Internal Error
VA_5	404 Not Found
VA_6	500 Server Internal Error
VA_7	500 Server Internal Error
VA_8	486 Busy Here
VA_9	480 Temporarily unavailable
VA_10	480 Temporarily unavailable
VA_11	480 Temporarily unavailable
VA_12	480 Temporarily unavailable
VA_13	410 Gone
VA_14	480 Temporarily unavailable
VA_15	502 Bad Gateway
VA_16	484 Address Incomplete
VA_17	500 Server Internal Error
VA_18	480 Temporarily unavailable
VA_19	486 Busy here if Diagnostics indicator includes the (CCBS indicator = CCBS possible) else 480 Temporarily unavailable
VA_20	500 Server Internal Error
VA_21	500 Server Internal Error
VA_22	500 Server Internal Error (SIP-I only)
VA_23	500 Server Internal Error
VA_24	500 Server Internal Error
VA_25	500 Server Internal Error
VA_26	500 Server Internal Error
VA_27	500 Server Internal Error
VA_28	500 Server Internal Error
VA_29	500 Server Internal Error
VA_30	404 Not Found
VA_31	500 Server Internal Error
VA_32	500 Server Internal Error
VA_33	500 Server Internal Error
VA_34	480 Temporarily unavailable
VA_35	500 Server Internal Error
VA_36	500 Server Internal Error
VA_37	500 Server Internal Error
VA_38	480 Temporarily unavailable

Table 6

Values for test purpose TP108002		
← SIP Message SIP_FAILURE_VA	← REL Cause Indicators parameter CV_ISUP,	
VA_1	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")
VA_2	480 Temporarily unavailable	Cause Value No. 18 ("No user responding")
VA_3	480 Temporarily unavailable	Cause Value No. 21 ("all rejected")
VA_4	410 Gone	Cause Value No. 22 ("number changed")
VA_5	502 Bad Gateway	Cause Value No. 27 ("destination out of order")
VA_6	484 Address Incomplete	Cause Value No. 28 ("invalid number format (address incomplete)")
VA_7	480 Temporarily unavailable	Cause Value No. 31 ("normal unspecified") (Class default)
VA_8	500 Server Internal Error	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)
VA_9	500 Server Internal Error	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_10	500 Server Internal Error	Cause Value No. 88 ("incompatible destination")
VA_11	500 Server Internal Error	Cause Value No. 111 ("protocol error, unspecified") (Class default)

TP108003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.2
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to " subscriber free ", having sent a 180 Ringing message on receipt of an ISUP REL: <ul style="list-style-type: none"> • The SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side. • The SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA. 	
SIP Parameter values:	SIP Status-Code: SIP_FAILURE_VA (PIXIT)	
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)	
Comments:	SIP INVITE 180 Ringing SIP_FAILURE_VA ACK	SUT → ← ← → → ISUP IAM ACM REL RLC

TP108004	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.2
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to " no indication ", having received a CPG message where the event information parameter event indicator is set to "Alerting", a 180 Ringing message is sent, on receipt of an ISUP REL: <ul style="list-style-type: none"> • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA. 	
SIP Parameter values:	SIP Status-Code: SIP_FAILURE_VA (PIXIT)	
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)	
Comments:	SIP INVITE 180 Ringing SIP_FAILURE_VA ACK	SUT → ← ← → → ISUP IAM ACM CPG REL RLC

Table 7

Values for test purposes TP108003 and TP108004		
← SIP Message SIP_FAILURE_VA		← REL Cause Indicators parameter CV_ISUP,
VA_1	480 Temporarily unavailable	Cause Value No. 21 ("all rejected")
VA_2	480 Temporarily unavailable	Cause Value No. 31 ("normal unspecified") (Class default)
VA_4	500 Server Internal Error	Cause Value No. 38 ("Network out of order")
VA_4	500 Server Internal Error	Cause Value No. 41 ("Temporary failure ")
VA_5	500 Server Internal Error	Cause Value No. 111 ("protocol error, unspecified") (Class default)

TP108005	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.2	
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/		
SIP selection criteria:	NOT PICS 4/10		
ISUP selection criteria:			
Test purpose:	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message, having received a ANM', a 200 OK message is sent, on receipt of an ISUP REL, where the cause value defined as CV_ISUP: <ul style="list-style-type: none"> the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; the SUT shall send a BYE message. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE 180 Ringing 200 OK INVITE	SUT → ← ← →	ISUP IAM ACM ANM
	BYE 200 OK BYE	← →	REL RLC

TP108006	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.2
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a CON message, a 200 OK message is sent, on receipt of an ISUP REL, where the cause value defined as CV_ISUP: <ul style="list-style-type: none"> • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send a BYE message. 	
SIP Parameter values:		
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)	
Comments:	SIP INVITE → 200 OK INVITE ← BYE ← 200 OK BYE →	SUT → ← IAM ← CON ← REL → RLC

Table 8

Values for test purpose TP108005 and TP 108006		
← SIP Message SIP_FAILURE_VA		← REL Cause Indicators parameter CV_ISUP,
VA_1	BYE	Cause Value No. 16
VA_2	BYE	Cause Value No. 31 ("normal unspecified") (Class default)
VA_3	BYE	Cause Value No. 38 ("Network out of order")
VA_4	BYE	Cause Value No. 41 ("Temporary failure ")
VA_5	BYE	Cause Value No. 111 ("protocol error, unspecified") (Class default)

TP108007	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.2
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, on receipt of an ISUP REL, where the cause value defined as CV_ISUP: <ul style="list-style-type: none"> • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field. 	
SIP Parameter values:	cause value: CV_SIP (PIXIT)	
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)	
Comments:	SIP INVITE SIP_FAILURE_VA ACK	SUT → ← → ISUP IAM REL RLC

Table 9

Values for test purposes TP108007		
	← SIP Message SIP_FAILURE_VA CV_SIP	← REL Cause Indicators parameter CV_ISUP,
VA_1	404 Not Found Cause Value No. 1	Cause Value No. 1 ("unallocated (unassigned) number")
VA_2	500 Server Internal Error Cause Value No. 2	Cause Value No. 2 ("no route to network")
VA_3	500 Server Internal Error Cause Value No. 3	Cause Value No. 3 ("no route to destination")
VA_4	500 Server Internal Error Cause Value No. 4	Cause Value No. 4 ("Send special information tone")
VA_5	404 Not Found Cause Value No. 5	Cause Value No. 5 ("Mis dialled trunk prefix")
VA_6	500 Server Internal Error Cause Value No. 8	Cause Value No. 8 ("Preemption")
VA_7	500 Server Internal Error Cause Value No. 9	Cause Value No. 9 ("Preemption-circuit reserved for reuse")
VA_8	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")
VA_9	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("no user responding")
VA_10	480 Temporarily unavailable Cause Value No. 19	Cause Value No. 19 ("no answer from the user")
VA_11	480 Temporarily unavailable Cause Value No. 20	Cause Value No. 20 ("subscriber absent")
VA_12	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")
VA_13	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")
VA_14	480 Temporarily unavailable Cause Value No. 25	Cause Value No. 25 ("Exchange routing error")
VA_15	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")
VA_16	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete)")
VA_17	500 Server Internal Error	Cause Value No. 29 ("facility rejected")
VA_18	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_19	486 Busy here if Diagnostics indicator includes the (CCBS indicator = CCBS possible) else 480 Temporarily unavailable Cause Value No. 34	Cause Value in the Class 010 (resource unavailable, Cause Value No. 34)
VA_20	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)
VA_21	500 Server Internal Error Cause Value No. 50	Cause Value No. 50 ("requested facility not subscribed")
VA_22	500 Server Internal Error Cause Value No. 55	Cause Value No. 55 ("incoming calls barred within CUG")
VA_23	500 Server Internal Error Cause Value No. 57	Cause Value No. 57 ("bearer capability not authorized")
VA_24	500 Server Internal Error Cause Value No. 58	Cause Value No. 58 ("bearer capability not presently")
VA_25	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_26	500 Server Internal Error Cause Value No. 65 - 79	Cause Value in the Class 100 (service or option not implemented Cause Value No. 65 - 79) (79 is class default)
VA_27	500 Server Internal Error Cause Value No. 87	Cause Value No. 87 ("user not member of CUG")

Values for test purposes TP108007		
← SIP Message SIP_FAILURE_VA CV_SIP		← REL Cause Indicators parameter CV_ISUP,
VA_28	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")
VA_29	500 Server Internal Error Cause Value No. 90	Cause Value No. 90 ("Non-existent CUG")
VA_30	404 Not Found Cause Value No. 91	Cause Value No. 91 ("invalid transit network selection")
VA_31	500 Server Internal Error Cause Value No. 95	Cause Value No. 95 ("invalid message") (Class default)
VA_32	500 Server Internal Error Cause Value No. 97	Cause Value No. 97 ("Message type non-existent or not implemented")
VA_33	500 Server Internal Error Cause Value No. 99	Cause Value No. 99 ("information element/parameter non-existent or not implemented")
VA_34	480 Temporarily unavailable Cause Value No. 102	Cause Value No. 102 ("recovery on timer expiry")
VA_35	500 Server Internal Error Cause Value No. 103	Cause Value No. 103 ("Parameter non-existent or not implemented, pass on")
VA_36	500 Server Internal Error Cause Value No. 110	Cause Value No. 110 ("Message with unrecognized Parameter, discarded")
VA_37	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)
VA_38	480 Temporarily unavailable Cause Value No. 127	Cause Value No. 127 ("interworking unspecified") (Class default)

TP108008	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.2
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to " no indication ", on receipt of an ISUP REL, where the cause value defined as CV_ISUP: <ul style="list-style-type: none"> • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field. 	
SIP Parameter values:	cause value: CV_SIP (PIXIT)	
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)	
Comments:	SIP INVITE SIP_FAILURE_VA ACK	SUT → ← → → ISUP IAM ACM REL RLC

Table 10

Values for test purpose TP108008		
← SIP Message SIP_FAILURE_VA CV_SIP		← REL Cause Indicators parameter CV_ISUP,
VA_1	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")
VA_2	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("No user responding")
VA_3	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")
VA_4	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")
VA_5	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")
VA_6	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete")
VA_7	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_8	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)
VA_9	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_10	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")
VA_11	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)

TP108009	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.2
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to " subscriber free ", having sent a 180 Ringing message on receipt of an ISUP REL, where the cause value defined as CV_ISUP: <ul style="list-style-type: none"> • the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; • the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; • the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field. 	
SIP Parameter values:	Cause value: CV_SIP (PIXIT)	
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)	
Comments:	SIP INVITE 180 Ringing SIP_FAILURE_VA ACK	SUT → ← ← → ISUP IAM ACM REL RLC

TP108010	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.2																									
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/																										
SIP selection criteria:	PICS 4/10																										
ISUP selection criteria:																											
Test purpose:	<p>Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message where the CPS indicator is set to "no indication", having received a CPG message where the event information parameter event indicator is set to "Alerting", a 180 Ringing message is sent, on receipt of an where the cause value defined as CV_ISUP:</p> <ul style="list-style-type: none"> the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; the SUT shall send the appropriate SIP status defined as SIP_FAILURE_VA; the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field. 																										
SIP Parameter values:	Cause value: CV_SIP (PIXIT)																										
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)																										
Comments:	<table style="width: 100%; text-align: center;"> <tr> <td>SIP INVITE</td> <td>→</td> <td>SUT</td> <td>→</td> <td>ISUP IAM</td> </tr> <tr> <td>180 Ringing</td> <td>←</td> <td></td> <td>←</td> <td>ACM</td> </tr> <tr> <td>SIP_FAILURE_VA</td> <td>←</td> <td></td> <td>←</td> <td>CPG</td> </tr> <tr> <td>ACK</td> <td>→</td> <td></td> <td>→</td> <td>REL</td> </tr> <tr> <td></td> <td></td> <td></td> <td></td> <td>RLC</td> </tr> </table>	SIP INVITE	→	SUT	→	ISUP IAM	180 Ringing	←		←	ACM	SIP_FAILURE_VA	←		←	CPG	ACK	→		→	REL					RLC	
SIP INVITE	→	SUT	→	ISUP IAM																							
180 Ringing	←		←	ACM																							
SIP_FAILURE_VA	←		←	CPG																							
ACK	→		→	REL																							
				RLC																							

Table 11

Values for test purposes TP108009 and TP108010		
←SIP Message SIP_FAILURE_VA CV_SIP		← REL Cause Indicators parameter CV_ISUP,
VA_1	486 Busy Here Cause Value No. 17	Cause Value No. 17 ("user busy")
VA_2	480 Temporarily unavailable Cause Value No. 18	Cause Value No. 18 ("No user responding")
VA_3	480 Temporarily unavailable Cause Value No. 21	Cause Value No. 21 ("all rejected")
VA_4	410 Gone Cause Value No. 22	Cause Value No. 22 ("number changed")
VA_5	502 Bad Gateway Cause Value No. 27	Cause Value No. 27 ("destination out of order")
VA_6	484 Address Incomplete Cause Value No. 28	Cause Value No. 28 ("invalid number format (address incomplete")
VA_7	480 Temporarily unavailable Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_8	500 Server Internal Error Cause Value No. 47	Cause Value in the Class 010 (resource unavailable, Cause Value No. 38-47) (47 is class default)
VA_9	500 Server Internal Error Cause Value No. 63	Cause Value No. 63 ("service option not available, unspecified") (Class default)
VA_10	500 Server Internal Error Cause Value No. 88	Cause Value No. 88 ("incompatible destination")
VA_11	500 Server Internal Error Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)

TP108011	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.2	
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/		
SIP selection criteria:	PICS 4/10		
ISUP selection criteria:			
Test purpose:	<p>Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, having received a ACM message, having received an ANM', a 200 OK message is sent, on receipt of an ISUP REL where the cause value defined as CV_ISUP:</p> <ul style="list-style-type: none"> the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; the SUT shall send BYE message; the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field in the BYE. 		
SIP Parameter values:	Cause value: CV_SIP (PIXIT)		
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)		
Comments:	SIP INVITE 180 Ringing 200 OK INVITE BYE 200 OK BYE	SUT → ← ← ← →	ISUP IAM ACM ANM REL RLC

TP108012	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.2	
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/		
SIP selection criteria:	PICS 4/10		
ISUP selection criteria:			
Test purpose:	<p>Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out a IAM message, having received a CON message, a 200 OK message is sent, on receipt of an where the cause value defined as CV_ISUP:</p> <ul style="list-style-type: none"> the SUT immediately requests the disconnection of the internal bearer path. When the ISUP circuit is available for re-selection, an ISUP RLC is returned to the ISUP side; the SUT shall send BYE message; the ISUP Cause Value field in the ISUP REL message is mapped to the Reason header field. 		
SIP Parameter values:	Cause value: CV_SIP (PIXIT)		
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)		
Comments:	SIP INVITE 200 OK INVITE BYE 200 OK BYE	SUT → ← ← → →	ISUP IAM CON REL RLC

Table 12

Values for test purposes TP108011 and TP108012		
\leftarrow SIP Message SIP_FAILURE_VA CV_SIP		\leftarrow REL Cause Indicators parameter CV_ISUP,
VA_1	BYE Cause Value No. 16	Cause Value No. 16
VA_2	BYE Cause Value No. 31	Cause Value No. 31 ("normal unspecified") (Class default)
VA_3	BYE Cause Value No. 38	Cause Value No. 38 ("Network out of order")
VA_4	BYE Cause Value No. 41	Cause Value No. 41 ("Temporary failure ")
VA_5	BYE Cause Value No. 111	Cause Value No. 111 ("protocol error, unspecified") (Class default)

TP108013	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.2
TSS reference:	SIP-ISUP /Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/10 AND PICS 4/21	
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT in the Idle state on receipt of a INVITE message, sending out an IAM message, on receipt of an ISUP REL with cause value 23 the SUT shall:</p> <ul style="list-style-type: none"> • the SUT immediately requests the redirection to the new destination according the ISUP/BICC procedures. 	
SIP Parameter values:		
ISUP Parameter values:	REL; cause value: 23	
Comments:	SIP INVITE	<p>SUT</p> <p>ISUP IAM REL RLC IAM</p>

6.2.1.9 Autonomous release at I-IWU

TP108101	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.3
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-IWU	
SIP selection criteria:		
ISUP selection criteria:	PICS 4/6	
Test purpose:	Ensure that when a an automatic repeat attempt initiated by the SUT is not successful (because the call is not routable), the SUT shall: <ul style="list-style-type: none"> send a 480 Temporarily Unavailable response to the SIP side. No actions on the ISUP (BICC) side are required. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP INVITE → 480 Temporarily unavailable ← ACK →	SUT → ← → ISUP IAM RSC RLC

TP108102	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.3
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-IWU	
SIP selection criteria:	NOT PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that when the SUT receives unrecognized backward ISUP or BICC signalling information and determines that the call needs to be released based on the coding of the message compatibility information, the SUT: <ul style="list-style-type: none"> shall send a 500 Server Internal Error response on the SIP side. 	
SIP Parameter values:		
ISUP Parameter values:	Unknown message: Message compatibility "Release call"	
Comments:	SIP INVITE → 180 Ringing ← → 500 Server internal error ← → ACK →	SUT → ← → ISUP IAM ACM ??? REL RLC

TP108103	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.3	
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-IWU		
SIP selection criteria:	PICS 4/10		
ISUP selection criteria:			
Test purpose:	Ensure that if the I-IWU receives unrecognized backward ISUP or BICC signalling information and determines that the call needs to be released based on the coding of the message compatibility information, the SUT shall: <ul style="list-style-type: none"> • send a 500 Server Internal Error response on the SIP side; • the reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value of the REL message sent by the I-IWU shall be contained in the SIP Message (BYE or final response) sent by the SIP side of the I-IWU. 		
SIP Parameter values:			
ISUP Parameter values:	Unknown message: Message compatibility "Release call"		
Comments:	SIP INVITE → 180 Ringing ← 500 Server internal error ← ACK →	SUT → ← ← → ←	ISUP IAM ACM ??? REL RLC

TP108104	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.3	
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-IWU		
SIP selection criteria:			
ISUP selection criteria:	PICS 3/4		
Test purpose:	Ensure that the SUT on receipt of insufficient digits received in an INVITE messages: <ul style="list-style-type: none"> • sends an 484 Address Incomplete message. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE → 484 Address incomplete ← ACK →	SUT → ← →	ISUP

TP108105	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.3															
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-IWU																
SIP selection criteria:	PICS 3/4																
ISUP selection criteria:																	
Test purpose:	<p>Ensure that the SUT on receipt of subsequent INVITE message:</p> <ul style="list-style-type: none"> • is sending a 484 Address Incomplete message to consider any offer-answer exchange initiated by the INVITE. A new INVITE shall initiate a new offer-answer exchange. 																
SIP Parameter values:																	
ISUP Parameter values:																	
Comments:	<p>As a general principle, the overlap procedures allow for session negotiation (and in particular the negotiation and confirmation of preconditions) to continue independently of the receipt of address information. On sending of a 484 Address Incomplete message for an INVITE transaction the I-IWU considers any offer-answer exchange initiated by the INVITE to be terminated. The new INVITE initiates a new offer-answer exchange. However, if resources have already been reserved and they can be reused within the new offer-answer exchange, the precondition signalling shall reflect the current status of the affected preconditions.</p> <table style="width: 100%; text-align: center;"> <tr> <td style="width: 33%;">SIP</td> <td style="width: 33%;">SUT</td> <td style="width: 33%;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td></td> </tr> <tr> <td>INVITE</td> <td>→</td> <td></td> </tr> <tr> <td>484 Address incomplete</td> <td>←</td> <td></td> </tr> <tr> <td>ACK</td> <td>→</td> <td></td> </tr> </table>		SIP	SUT	ISUP	INVITE	→		INVITE	→		484 Address incomplete	←		ACK	→	
SIP	SUT	ISUP															
INVITE	→																
INVITE	→																
484 Address incomplete	←																
ACK	→																

TP108106	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.3												
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-IWU													
SIP selection criteria:														
ISUP selection criteria:														
Test purpose:	<p>Ensure that the SUT in congestion on receipt of INVITE message:</p> <ul style="list-style-type: none"> • sends an 480 Temporarily Unavailable message. 													
SIP Parameter values:														
ISUP Parameter values:														
Comments:	<table style="width: 100%; text-align: center;"> <tr> <td style="width: 33%;">SIP</td> <td style="width: 33%;">SUT</td> <td style="width: 33%;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td></td> </tr> <tr> <td>480 Temporarily unavailable</td> <td>←</td> <td></td> </tr> <tr> <td>ACK</td> <td>→</td> <td></td> </tr> </table>		SIP	SUT	ISUP	INVITE	→		480 Temporarily unavailable	←		ACK	→	
SIP	SUT	ISUP												
INVITE	→													
480 Temporarily unavailable	←													
ACK	→													

TP108107	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.3	
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-IWU		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	Ensure that the release procedure of the BICC/ISUP are a result of a release after answer :		
	<ul style="list-style-type: none"> • sends a BYE message to the UAC; • sends a REL to the BICC/ISUP side. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← BYE ← 200 OK BYE →	SUT → ← ← → ←	ISUP IAM ACM ANM REL RLC

TP108108	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.3	
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-IWU		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	Ensure that the call is released due to the BICC/ISUP compatibility procedure for unknown parameters:		
	<ul style="list-style-type: none"> • sends 500 Server Internal Error. 		
SIP Parameter values:			
ISUP Parameter values:	Unknown parameter in ACM: Parameter compatibility "Release call"		
Comments:	SIP INVITE → 500 Server internal error ← ACK →	SUT → → →	ISUP IAM ACM(???) REL RLC

TP108109	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.3	
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-IWU		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	Ensure that the call is released due to expiry of T7 within the BICC/ISUP procedures: <ul style="list-style-type: none"> • sends 484 Address Incomplete. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE → 484 Address incomplete ← ACK →	SUT T7 expiry → REL ←	ISUP IAM → REL ←

TP108110	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.3	
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-IWU		
SIP selection criteria:			
ISUP selection criteria:	PICS 4/16		
Test purpose:	Ensure that the call is released due expiry of T9 within the BICC/ISUP procedures: <ul style="list-style-type: none"> • sends 480 Temporarily Unavailable. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE → 180 Ringing ← 480 Temporarily unavailable ← ACK →	SUT T9 expiry → REL ←	ISUP IAM → ACM ← REL ←

TP108111	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.3	
TSS reference:	SIP-ISUP/Basic call/ Autonomous release at I-IWU		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	Ensure that the call is released due release before answer: <ul style="list-style-type: none">• sends 480 Temporarily Unavailable.		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE → 180 Ringing ← → Autonomous release from I-IWU 480 Temporarily unavailable ← → ACK → ←	SUT → IAM ACM REL RLC	ISUP IAM ACM REL RLC

6.2.1.10 Receipt of the Release message BYE / CANCEL

TP109001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.1	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the BYE message		
SIP selection criteria:	NOT PICS 4/11		
ISUP selection criteria:			
Test purpose:	Ensure that the SUT on receipt of SIP BYE , the SUT shall send an ISUP REL with the cause value # 16 to the ISUP side.		
SIP Parameter values:			
ISUP Parameter values:	REL: Cause value #16, Location "Network beyond an interworking point"		
Comments:	SIP INVITE → 180 Ringing ← → 200 OK INVITE ← → BYE → 200 OK BYE ←	SUT → IAM ACM ANM REL RLC	ISUP IAM ACM ANM REL RLC

TP109002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.1	
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CANCEL message		
SIP selection criteria:	NOT PICS 4/11		
ISUP selection criteria:			
Test purpose:	Ensure that the SUT on receipt of SIP CANCEL , the I-IWU shall send an ISUP REL with the cause value # 31 to the ISUP side.		
SIP Parameter values:			
ISUP Parameter values:	REL: Cause value #31, Location "Network beyond an interworking point"		
Comments:	SIP INVITE → 180 Ringing ← → CANCEL → 200 OK CANCEL ←	SUT → IAM ACM REL RLC	ISUP IAM ACM REL RLC

TP109003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.1																		
TSS reference:	SIP-ISUP/Basic call/ Receipt of the BYE message																			
SIP selection criteria:	PICS 4/11																			
ISUP selection criteria:																				
Test purpose:	<p>Ensure that the SUT on receipt of SIP BYE, the SUT shall send an ISUP REL to the ISUP side.</p> <p>Ensure that the Reason header field with ITU-T Rec Q.850 [5] Cause Value is included in the BYE message is mapped to the ISUP Cause Value field in the ISUP REL message.</p>																			
SIP Parameter values:	Protocol-cause: CV_Reason Header (PIXIT)																			
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT)																			
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 33%;">SIP</td> <td style="width: 33%; text-align: center;">SUT</td> <td style="width: 33%;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td>→</td> </tr> <tr> <td>180 Ringing</td> <td>←</td> <td>←</td> </tr> <tr> <td>200 OK INVITE</td> <td>←</td> <td>←</td> </tr> <tr> <td>BYE</td> <td>→</td> <td>→</td> </tr> <tr> <td>200 OK BYE</td> <td>←</td> <td>←</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	→	180 Ringing	←	←	200 OK INVITE	←	←	BYE	→	→	200 OK BYE	←	←	
SIP	SUT	ISUP																		
INVITE	→	→																		
180 Ringing	←	←																		
200 OK INVITE	←	←																		
BYE	→	→																		
200 OK BYE	←	←																		

TP109004	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.1															
TSS reference:	SIP-ISUP/Basic call/ Receipt of the CANCEL message																
SIP selection criteria:	PICS 4/11																
ISUP selection criteria:																	
Test purpose:	<p>Ensure that the SUT on receipt of SIP CANCEL, the I-IWU shall send an ISUP REL to the ISUP side.</p> <p>Ensure that the Reason header field with ITU-T Rec Q.850 [5] Cause Value is included in the CANCEL message is mapped to the ISUP Cause Value field in the ISUP REL message.</p>																
SIP Parameter values:																	
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT) location: LOC_ISUP (PIXIT)																
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 33%;">SIP</td> <td style="width: 33%; text-align: center;">SUT</td> <td style="width: 33%;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td>→</td> </tr> <tr> <td>180 Ringing</td> <td>←</td> <td>←</td> </tr> <tr> <td>CANCEL</td> <td>→</td> <td>→</td> </tr> <tr> <td>200 OK CANCEL</td> <td>←</td> <td>←</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	→	180 Ringing	←	←	CANCEL	→	→	200 OK CANCEL	←	←	
SIP	SUT	ISUP															
INVITE	→	→															
180 Ringing	←	←															
CANCEL	→	→															
200 OK CANCEL	←	←															

6.2.1.11 Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented

TP110001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.11.4 and 5			
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria:					
ISUP selection criteria:					
Test purpose:	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a RSC message sends: <ul style="list-style-type: none"> • a BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent. 				
SIP Parameter values:					
ISUP Parameter values:					
Comments:	SIP INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK BYE	SUT → ← ← → ← →	ISUP IAM ACM ANM RSC RLC		

TP110002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.11.4 and 5			
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented				
SIP selection criteria:					
ISUP selection criteria:					
Test purpose:	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a GRS message sends: <ul style="list-style-type: none"> • a BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent. 				
SIP Parameter values:					
ISUP Parameter values:					
Comments:	SIP INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK BYE	SUT → ← ← → ← →	ISUP IAM ACM ANM GRS GRA		

TP110003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.4																					
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																						
SIP selection criteria:																							
ISUP selection criteria:																							
Test purpose:	<p>Ensure that the SUT, when at least one backward ISUP message relating to the call has already been received on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented", sends:</p> <ul style="list-style-type: none"> • a BYE message if the SUT has already received an ACK for the 200 OK INVITE message which had it sent. 																						
SIP Parameter values:																							
ISUP Parameter values:	Circuit Group Supervision Message Type Indicator "hardware failure oriented"																						
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP</th> <th style="text-align: center; width: 40%;">SUT</th> <th style="text-align: right; width: 30%;">ISUP</th> </tr> </thead> <tbody> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">IAM</td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: right;">ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">ANM</td> </tr> <tr> <td>ACK</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>BYE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">CGB</td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">CGBA</td> </tr> </tbody> </table>		SIP	SUT	ISUP	INVITE	→	IAM	180 Ringing	←	ACM	200 OK INVITE	←	ANM	ACK	→		BYE	←	CGB	200 OK BYE	→	CGBA
SIP	SUT	ISUP																					
INVITE	→	IAM																					
180 Ringing	←	ACM																					
200 OK INVITE	←	ANM																					
ACK	→																						
BYE	←	CGB																					
200 OK BYE	→	CGBA																					

TP110004	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.11.4 and 5																					
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																						
SIP selection criteria:																							
ISUP selection criteria:																							
Test purpose:	<p>Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a RSC message sends 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE.</p> <ul style="list-style-type: none"> • The SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE. 																						
SIP Parameter values:																							
ISUP Parameter values:																							
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SIP	SUT	ISUP																					
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200 OK INVITE	←	ANM																					
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BYE	←	RLC																					
200 OK BYE	→																						

TP110005	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.11.4 and 5	
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	<p>Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a GRS message sends 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE.</p> <ul style="list-style-type: none"> The SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← ACK → BYE ← 200 OK BYE →	SUT → ← ← ← → → →	ISUP IAM ACM ANM GRS GRA

TP110006	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.4	
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	<p>Ensure that the SUT, when at least one backward ISUP message relating to the call has already been received on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented", sends 200 OK INVITE if the SUT has not yet received an ACK for the 200 OK INVITE.</p> <ul style="list-style-type: none"> The SUT shall wait until it receives the ACK for the 200 OK INVITE before sending the BYE. 		
SIP Parameter values:			
ISUP Parameter values:	Circuit Group Supervision Message Type Indicator "hardware failure oriented"		
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← ACK → BYE ← 200 OK BYE →	SUT → ← ← ← → → →	ISUP IAM ACM ANM CGB GGBA

TP110007	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.11.4 and 5															
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																
SIP selection criteria:																	
ISUP selection criteria:																	
Test purpose:	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a RSC message sends: <ul style="list-style-type: none"> a 500 Server Internal Error on the SIP side. 																
SIP Parameter values:																	
ISUP Parameter values:																	
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 33%;">SIP</td> <td style="width: 33%; text-align: center;">SUT</td> <td style="width: 33%;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td>IAM</td> </tr> <tr> <td>180 Ringing</td> <td>←</td> <td>ACM</td> </tr> <tr> <td>500 Server Internal Error</td> <td>←</td> <td>RSC</td> </tr> <tr> <td>ACK</td> <td>→</td> <td>RLC</td> </tr> </table>		SIP	SUT	ISUP	INVITE	→	IAM	180 Ringing	←	ACM	500 Server Internal Error	←	RSC	ACK	→	RLC
SIP	SUT	ISUP															
INVITE	→	IAM															
180 Ringing	←	ACM															
500 Server Internal Error	←	RSC															
ACK	→	RLC															

TP110008	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.11.4 and 5															
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																
SIP selection criteria:																	
ISUP selection criteria:																	
Test purpose:	Ensure that the SUT, when at least one backward ISUP/BICC message relating to the call has already been received on receipt of a GRS message sends: <ul style="list-style-type: none"> a 500 Server Internal Error on the SIP side. 																
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ISUP Parameter values:																	
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SIP	SUT	ISUP															
INVITE	→	IAM															
180 Ringing	←	ACM															
500 Server Internal Error	←	GRS															
ACK	→	GRA															

TP110009	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.11.4	
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	<p>Ensure that the SUT, when at least one backward ISUP message relating to the call has already been received on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented", sends:</p> <ul style="list-style-type: none"> • a 500 Server Internal Error on the SIP side. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE → 180 Ringing ← → 500 Server Internal Error ← → ACK ← →	SUT → → ← ← → →	ISUP IAM ACM CGB CGBA

TP110010	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.11.4 and 5	
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	<p>Ensure that the SUT after receiving more than one INVITE sending an IAM message for each call association on receipt of a GRS message where the Range and Status Parameter value is bigger than "1":</p> <ul style="list-style-type: none"> • the SUT shall send a BYE requests for each call association. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE 1 → 180 Ringing ← → 200 OK INVITE ← → ACK → INVITE 2 → 180 Ringing ← → 200 OK INVITE ← → ACK → BYE 1 ← → 200 OK BYE → → BYE 2 ← → 200 OK BYE →	SUT → → → → → → → → → → → → → → → → → → → → → → → →	ISUP IAM ACM ANM IAM ACM ANM GRS GRA

TP110011	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 6.11.4 and 5																																													
TSS reference:	SIP-ISUP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																																														
SIP selection criteria:																																															
ISUP selection criteria:																																															
Test purpose:	Ensure that the SUT after receiving more than one INVITE sending an IAM message for each call association on receipt of a CGB message, with the Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" where the Range and Status Parameter value is bigger than "1" . <ul style="list-style-type: none"> • the SUT shall send a BYE requests for each call association. 																																														
SIP Parameter values:																																															
ISUP Parameter values:																																															
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">SIP</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">ISUP</td> </tr> <tr> <td>INVITE 1</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> <tr> <td>ACK</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td> </td> <td></td> <td></td> </tr> <tr> <td>INVITE 2</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> <tr> <td>ACK</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td> </td> <td></td> <td></td> </tr> <tr> <td>BYE 1</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> </tr> <tr> <td>BYE 2</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">→</td> <td></td> </tr> </table>	SIP	SUT	ISUP	INVITE 1	→	→	180 Ringing	←	←	200 OK INVITE	←	←	ACK	→					INVITE 2	→	→	180 Ringing	←	←	200 OK INVITE	←	←	ACK	→					BYE 1	←	←	200 OK BYE	→	→	BYE 2	←		200 OK BYE	→		
SIP	SUT	ISUP																																													
INVITE 1	→	→																																													
180 Ringing	←	←																																													
200 OK INVITE	←	←																																													
ACK	→																																														
INVITE 2	→	→																																													
180 Ringing	←	←																																													
200 OK INVITE	←	←																																													
ACK	→																																														
BYE 1	←	←																																													
200 OK BYE	→	→																																													
BYE 2	←																																														
200 OK BYE	→																																														

6.2.1.12 Receipt of the Suspend message (SUS) network initiated

TP111001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 6.9															
TSS reference:	SIP-ISUP/Basic call/ receipt of a SUSPEND message with the suspend indicator set to "network initiated"																
SIP selection criteria:																	
ISUP selection criteria:																	
Test purpose:	Ensure that the SUT, on receipt of a SUSPEND message with the suspend indicator set to "network initiated": <ul style="list-style-type: none"> • does not send any message. 																
SIP Parameter values:																	
ISUP Parameter values:	SUS; Suspend indicator : network initiated																
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">SIP</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">ISUP</td> </tr> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> <tr> <td></td> <td></td> <td style="text-align: center;">SUS</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	→	180 Ringing	←	←	200 OK INVITE	←	←			SUS	
SIP	SUT	ISUP															
INVITE	→	→															
180 Ringing	←	←															
200 OK INVITE	←	←															
		SUS															

6.2.1.13 Receipt of the Resume message (RES) network initiated

TP112001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1] clause 6.10
TSS reference:	SIP-ISUP/Basic call/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT, on receipt of a RESUME message containing the suspend/resume indicator set to "network initiated": <ul style="list-style-type: none">• does not send any message.	
SIP Parameter values:		
ISUP Parameter values:	RES; Suspend indicator: network initiated	
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ←	SUT → IAM ← ACM ← ANM ← SUS ← RES ←

6.2.2 Interworking from ISUP to SIP

6.2.2.1 Sending of the INVITE message

TP301001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, 1 a)
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication:	
	<ul style="list-style-type: none"> • sends the INVITE message. 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number: with sending complete indication	
Comments:	ISUP/BICC IAM	SUT → SIP INVITE

TP301002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, 1 b)
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan:	
	<ul style="list-style-type: none"> • sends the INVITE message. 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number: complete number	
Comments:	ISUP/BICC IAM	SUT → SIP INVITE

TP301003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, 1 c)
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party:	
	<ul style="list-style-type: none"> • sends the INVITE message. 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number: complete number	
Comments:	ISUP/BICC IAM	SUT → SIP INVITE

TP301004	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, 1 d)
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{oiw1} after the receipt of the latest address message: <ul style="list-style-type: none"> • sends the INVITE message. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP/BICC IAM	SUT \rightarrow T_{oiw1} expiry \rightarrow SIP INVITE

TP301005	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, A)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:	PICS 1/5	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message with the complete called party number containing the Continuity Check indicator in the Nature of Connection Indicators parameter is set to indicate " continuity check not required ": <ul style="list-style-type: none"> • sends a INVITE message. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM	SUT \rightarrow \rightarrow SIP INVITE

TP301006	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, A)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:	PICS 1/5	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message with the complete called party number containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to " continuity check required on this circuit ":	
	<ul style="list-style-type: none"> sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "continuity check successful". 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM COT	SUT → → → SIP INVITE

TP301007	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, A)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:	PICS 1/5	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message with the complete called party number containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to " continuity check performed on previous circuit ":	
	<ul style="list-style-type: none"> sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "continuity check successful". 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM COT	SUT → → → SIP INVITE

TP301008	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, A)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to " continuity check required on this circuit ". INVITE shall not be sent if the Continuity message is received with the Continuity Indicators parameter set to " continuity check failed ".	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM COT	SUT → → SIP

TP301009	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, A)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:	PICS 1/5	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to " continuity check required on this circuit ". INVITE shall not be sent if the ISUP timer T8 expires. The SUT: <ul style="list-style-type: none"> • sends a REL message. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM REL RLC	SUT → T8 expiry ← → SIP

TP301012	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, B)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/5 AND PICS 4/2	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate " continuity check not required ": <ul style="list-style-type: none"> • sends an INVITE message with precondition using the SDP offer in the INVITE. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM	SUT → INVITE → SIP

TP301013	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, B)	
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message		
SIP selection criteria:	PICS 4/5 AND PICS 4/15		
ISUP selection criteria:	PICS 1/5 AND PICS 4/2		
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate " continuity check required on this circuit ": <ul style="list-style-type: none"> • sends an INVITE message with precondition using the SDP offer in the INVITE. The SDP offer or answer carrying the confirmation of a precondition being met is sent when the Continuity message with the Continuity Indicators parameter set to "continuity check successful" was received and the requested preconditions are met in the SIP network. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP IAM → COT →	SUT → ← → ← → ←	SIP INVITE 183 Session Progress PRACK 200 OK PRACK UPDATE 200 OK UPDATE

TP301014	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, B)	
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message		
SIP selection criteria:	PICS 4/5 AND PICS 4/15		
ISUP selection criteria:	PICS 1/5 AND PICS 4/2		
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message where the Continuity Check indicator in the Nature of Connection Indicators parameter in the IAM is set to indicate " continuity check performed on previous circuit ": <ul style="list-style-type: none"> • sends an INVITE message with precondition using the SDP offer in the INVITE. The SDP offer or answer carrying the confirmation of a precondition being met is sent when the Continuity message with the Continuity Indicators parameter set to "continuity check successful" was received and the requested preconditions are met in the SIP network. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP IAM → COT →	SUT → ← → ← → ←	SIP INVITE 183 Session Progress PRACK 200 OK PRACK UPDATE 200 OK UPDATE

TP301015	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, B)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/5 AND PICS 4/2	
Test purpose:	<p>The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check performed on previous circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The call has been cleared before an early dialogue has been established. Ensure that the SUT:</p> <ul style="list-style-type: none"> • sends CANCEL if on the SIP side the internal resource reservation was unsuccessful; • REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<p>ISUP IAM → SUT ← SIP INVITE internal resource reservation was unsuccessful REL ← → CANCEL RLC → ← 200 OK CANCEL</p>	

TP301016	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, B)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/5 AND PICS 4/2	
Test purpose:	<p>The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check performed on previous circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established. Ensure that the SUT:</p> <ul style="list-style-type: none"> • sends CANCEL if on the SIP side the internal resource reservation was unsuccessful; • REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<p>ISUP IAM → SUT ← SIP INVITE internal resource reservation was unsuccessful REL ← → CANCEL RLC → ← 200 OK CANCEL terminated → ← 487 Request → ACK</p>	

TP301019	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, B)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/5 AND PICS 4/2	
Test purpose:	<p>The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The call has been cleared before an early dialogue has been established. Ensure that the SUT:</p> <ul style="list-style-type: none"> Sends CANCEL if on the SIP side the internal resource reservation was unsuccessful. REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<p>ISUP IAM → SUT → ↗ ↙ SIP INVITE 100 Trying</p> <p>internal resource reservation was unsuccessful</p> <p>REL ↙ → CANCEL RLC ↗ ↙ 200 OK CANCEL</p>	

TP301020	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, B)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/5 AND PICS 4/2	
Test purpose:	<p>The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established. Ensure that the SUT:</p> <ul style="list-style-type: none"> • Sends CANCEL if on the SIP side the internal resource reservation was unsuccessful. • REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<p>ISUP IAM</p> <pre> sequenceDiagram participant SUT participant IWU SUT->>IWU: IAM IWU->>SUT: SIP INVITE SUT->>IWU: CANCEL IWU->>SUT: 200 OK CANCEL SUT->>IWU: ACK note over SUT: internal resource reservation was unsuccessful </pre>	<p>SIP INVITE SIP_MESSAGE_VA</p> <p>internal resource reservation was unsuccessful</p> <p>CANCEL 200 OK CANCEL 487 Request terminated ACK</p>

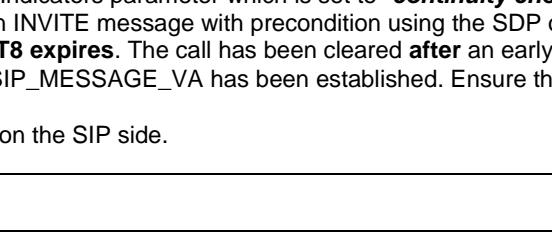
Table 13

Values for test purpose: TP301016, TP301020, TP3010128, TP3010132 and TP3010148	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress without SDP

TP301027	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, B)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/5 AND PICS 4/2	
Test purpose:	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to " continuity check required on this circuit " and sends an INVITE message with precondition using the SDP offer in the INVITE. The Continuity message is received with the Continuity Indicators parameter set to " continuity check failed ". The call has been cleared before an early dialogue has been established. Ensure that the SUT: <ul style="list-style-type: none"> sends CANCEL on the SIP side. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM → COT →	SUT → ← 100 Trying → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK

TP301028	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, B)	
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message		
SIP selection criteria:	PICS 4/5 AND PICS 4/15		
ISUP selection criteria:	PICS 1/5 AND PICS 4/2		
Test purpose:	<p>The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The Continuity message is received with the Continuity Indicators parameter set to "continuity check failed". The call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established. Ensure that the SUT:</p> <ul style="list-style-type: none"> sends CANCEL on the SIP side. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP IAM → COT →	SUT → ← → ← →	SIP INVITE SIP_MESSAGE_VA CANCEL 200 OK CANCEL 487 Request terminated ACK

TP301031	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, B)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/5 AND PICS 4/2	
Test purpose:	<p>The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" and sends an INVITE message with precondition using the SDP offer in the INVITE. The ISUP Timer T8 expires. The call has been cleared before an early dialogue has been established. Ensure that the SUT:</p> <ul style="list-style-type: none"> sends CANCEL on the SIP side. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<p>ISUP IAM → SUT → SIP INVITE REL RLC ← T8 expires ← 100 Trying → → CANCEL ← ← 200 OK CANCEL ← ← 487 Request terminated → → ACK</p>	

TP301032	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, B)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/5 AND PICS 4/2	
Test purpose:	The SUT in Idle state, receives an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to " continuity check required on this circuit " and sends an INVITE message with precondition using the SDP offer in the INVITE. The ISUP Timer T8 expires . The call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established. Ensure that the SUT: <ul style="list-style-type: none"> sends CANCEL on the SIP side. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM REL RLC	SUT  SIP INVITE SIP_MESSAGE_VA CANCEL 200 OK CANCEL 487 Request terminated ACK

TP301037	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, C)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:	PICS 1/4	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message indicating " COT to be expected ": <ul style="list-style-type: none"> • The sending of the INVITE is delayed until all the following conditions are satisfied: <ul style="list-style-type: none"> - Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received; - Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	BICC IAM COT	SUT → → → SIP INVITE

TP301038	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, C)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:	PICS 1/4	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message indicating " COT to be expected ": <ul style="list-style-type: none"> • The sending of the INVITE is delayed until all the following conditions are satisfied: <ul style="list-style-type: none"> - Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received; - APM with Action indicator set to "Connected" - for the forward bearer set-up cases (with, or without bearer control tunnelling) where the incoming Connect Type is "notification required", and for the fast set-up (backward) case. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	BICC IAM COT APM	SUT → → → SIP INVITE

TP301039	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, C)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:	PICS 1/4	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message indicating " COT to be expected ": <ul style="list-style-type: none"> • The sending of the INVITE delays until all the following conditions are satisfied: <ul style="list-style-type: none"> - Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received; - Bearer Set-up Connect indication - for the backward bearer set-up case was received. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	BICC IAM COT	SUT → → → SIP INVITE

TP301040	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, C) 2.4
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:	PICS 1/4	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message indicating " COT to be expected ": <ul style="list-style-type: none"> • The sending of the INVITE delays until all the following conditions are satisfied: <ul style="list-style-type: none"> - Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received; - BNC set-up success indication for cases using bearer control tunnelling was received. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	BICC IAM COT	SUT → → → SIP INVITE

TP301041	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, C)																
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message																	
SIP selection criteria:	NOT PICS 4/15																	
ISUP selection criteria:	PICS 1/4																	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected":</p> <ul style="list-style-type: none"> • Sends not the INVITE if the Continuity message was not received, i.e., the BICC timer T8 expires. - Send REL with Cause Value 41 (<i>temporary failure</i>) shall be sent on the BICC side of the O-IWU. 																	
SIP Parameter values:																		
ISUP Parameter values:																		
Comments:	<table style="width: 100%; text-align: center;"> <tr> <td>BICC</td> <td></td> <td>SUT</td> <td>SIP</td> </tr> <tr> <td>IAM</td> <td>→</td> <td></td> <td></td> </tr> <tr> <td>REL</td> <td>←</td> <td>T8 expires</td> <td></td> </tr> <tr> <td>RLC</td> <td>→</td> <td></td> <td></td> </tr> </table>	BICC		SUT	SIP	IAM	→			REL	←	T8 expires		RLC	→			
BICC		SUT	SIP															
IAM	→																	
REL	←	T8 expires																
RLC	→																	

TP301042	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, D)																				
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message																					
SIP selection criteria:	PICS 4/5 AND PICS 4/15																					
ISUP selection criteria:	PICS 1/4 AND PICS 4/2																					
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected" sends an INVITE message with precondition using the SDP offer in the INVITE. The precondition signalling is concluded upon sending the (within an SDP offer-answer exchange) confirmation of a precondition being met. The SDP offer or answer carrying the confirmation of a precondition being met is sent when all of the following conditions are satisfied when:</p> <ul style="list-style-type: none"> • Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received; • bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received. 																					
SIP Parameter values:																						
ISUP Parameter values:																						
Comments:	<table style="width: 100%; text-align: center;"> <tr> <td>BICC</td> <td></td> <td>SUT</td> <td>SIP</td> </tr> <tr> <td>IAM</td> <td>→</td> <td>→</td> <td>INVITE</td> </tr> <tr> <td>COT</td> <td></td> <td>←</td> <td>183 Session Progress</td> </tr> <tr> <td></td> <td></td> <td>→</td> <td>UPDATE</td> </tr> <tr> <td></td> <td></td> <td>←</td> <td>200 OK UPDATE</td> </tr> </table>	BICC		SUT	SIP	IAM	→	→	INVITE	COT		←	183 Session Progress			→	UPDATE			←	200 OK UPDATE	
BICC		SUT	SIP																			
IAM	→	→	INVITE																			
COT		←	183 Session Progress																			
		→	UPDATE																			
		←	200 OK UPDATE																			

TP301043	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, D) 2.2
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/4 AND PICS 4/2	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected" sends an INVITE message with precondition using the SDP offer in the INVITE. The precondition signalling is concluded upon sending the (within an SDP offer-answer exchange) confirmation of a precondition being met. The SDP offer or answer carrying the confirmation of a precondition being met is sent when all of the following conditions are satisfied when:</p> <ul style="list-style-type: none"> • continuity message, with the Continuity Indicators parameter set to "continuity" shall be received; • APM with Action indicator set to "Connected" - for the forward bearer set-up cases (with, or without bearer control tunnelling) where the incoming Connect Type is "notification required", and for the fast set-up (backward) case. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	BICC IAM COT	<p style="text-align: center;">SUT</p> <pre> sequenceDiagram participant SUT participant SIP SUT->>SIP: IAM activate SIP SIP-->>SUT: 183 Session Progress deactivate SIP SUT->>SIP: COT activate SIP SIP-->>SUT: 200 OK UPDATE deactivate SIP </pre>

TP301044	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, D) 2.3
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	NOT PICS 4/15	
ISUP selection criteria:	PICS 1/4	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected" sends an INVITE message with precondition using the SDP offer in the INVITE. The precondition signalling is concluded upon sending the (within an SDP offer-answer exchange) confirmation of a precondition being met. The SDP offer or answer carrying the confirmation of a precondition being met is sent when all of the following conditions are satisfied when:</p> <ul style="list-style-type: none"> • Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received; • Bearer Set-up Connect indication - for the backward bearer set-up case was received. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	BICC IAM COT	<p style="text-align: center;">SUT</p> <pre> sequenceDiagram participant SUT participant SIP SUT->>SIP: IAM activate SIP SIP-->>SUT: 183 Session Progress deactivate SIP SUT->>SIP: COT activate SIP SIP-->>SUT: 200 OK UPDATE deactivate SIP </pre>

TP301045	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, D) 2.4
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/4 AND PICS 4/2	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected" sends an INVITE message with precondition using the SDP offer in the INVITE. The precondition signalling is concluded upon sending the (within an SDP offer-answer exchange) confirmation of a precondition being met. The SDP offer or answer carrying the confirmation of a precondition being met is sent when all of the following conditions are satisfied when:</p> <ul style="list-style-type: none"> • Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received; • BNC set-up success indication for cases using bearer control tunnelling was received. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	BICC IAM COT	<pre> sequenceDiagram participant SUT participant ISUP SUT->>ISUP: SIP INVITE activate ISUP ISUP->>SUT: 183 Session Progress UPDATE deactivate ISUP SUT->>ISUP: SIP 200 OK UPDATE </pre>

TP301046	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, D)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/4 AND PICS 4/2	
Test purpose:	The SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to " COT to be expected ", sends an INVITE message with precondition using the SDP offer in the INVITE:	<ul style="list-style-type: none"> • Ensure that the SUT sends CANCEL if the ISUP timer T8 expires if the call has been cleared before an early dialogue has been established.
SIP Parameter values:		
ISUP Parameter values:		
Comments:	BICC IAM REL RLC	<pre> sequenceDiagram participant SUT participant ISUP SUT->>ISUP: SIP INVITE activate ISUP ISUP->>SUT: 100 Trying deactivate ISUP SUT->>ISUP: CANCEL activate ISUP ISUP->>SUT: 200 OK CANCEL </pre>

TP301048	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1, D)
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/4 AND PICS 4/2	
Test purpose:	The SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to " COT to be expected ", sends an INVITE message with precondition using the SDP offer in the INVITE: <ul style="list-style-type: none"> Ensure that the SUT sends CANCEL if the ISUP timer T8 expires if the call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	BICC IAM REL RLC	<p style="text-align: center;">SUT</p> <pre> graph LR IAM[BICC IAM] --> INVITE[SIP INVITE] INVITE --> SIPMESSAGE[SIP MESSAGE_VA] T8[T8 expires] --> CANCEL[SIP CANCEL] CANCEL --> ACK[SIP ACK] CANCEL --> CANCEL2[SIP 200 OK CANCEL] CANCEL2 --> ACK2[SIP ACK] CANCEL2 --> REQUEST487[SIP 487 Request terminated] REQUEST487 --> ACK3[SIP ACK] IAM --> TRYING[ISUP 100 Trying] TRYING --> CANCEL_ISUP[SIP CANCEL] CANCEL_ISUP --> ACK_ISUP[SIP ACK] CANCEL_ISUP --> CANCEL2_ISUP[SIP 200 OK CANCEL] CANCEL2_ISUP --> REQUEST487_ISUP[SIP 487 Request terminated] REQUEST487_ISUP --> ACK4[SIP ACK] </pre>

TP301049	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/4 AND PICS 4/2	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing Continuity Check indicator in the Nature of Connection Indicators parameter which is set to " COT to be expected ". Ensure that the SUT: <ul style="list-style-type: none"> Sends CANCEL if on the SIP side the internal resource reservation was unsuccessful and if the call has been cleared before an early dialogue with the message has been established. A REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	BICC IAM REL RLC	<p style="text-align: center;">SUT</p> <pre> graph LR IAM[BICC IAM] --> INVITE[SIP INVITE] INVITE --> TRYING[SIP 100 Trying] TRYING --> CANCEL[SIP CANCEL] CANCEL --> ACK[SIP ACK] CANCEL --> CANCEL2[SIP 200 OK CANCEL] CANCEL2 --> REQUEST487[SIP 487 Request terminated] REQUEST487 --> ACK2[SIP ACK] IAM --> TRYING_ISUP[ISUP 100 Trying] TRYING_ISUP --> CANCEL_ISUP[SIP CANCEL] CANCEL_ISUP --> ACK_ISUP[SIP ACK] CANCEL_ISUP --> CANCEL2_ISUP[SIP 200 OK CANCEL] CANCEL2_ISUP --> REQUEST487_ISUP[SIP 487 Request terminated] REQUEST487_ISUP --> ACK4[SIP ACK] </pre>

TP301051	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1												
TSS reference:	ISUP-SIP/Basic call/Sending of the INVITE message													
SIP selection criteria:	PICS 4/5 AND PICS 4/15													
ISUP selection criteria:	PICS 1/4 AND PICS 4/2													
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "COT to be expected". Ensure that the SUT:</p> <ul style="list-style-type: none"> • Sends CANCEL if on the SIP side the internal resource reservation was unsuccessful and if the call has been cleared after an early dialogue with the message defined as SIP_MESSAGE_VA has been established. • A REL with Cause Value 47 (resource unavailable, unspecified) shall be sent on the ISUP side of the O-IWU. 													
SIP Parameter values:														
ISUP Parameter values:														
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">BICC IAM</td> <td style="width: 50%; text-align: center;">SUT</td> <td style="width: 25%;">SIP INVITE SIP_MESSAGE_VA</td> </tr> <tr> <td></td> <td style="text-align: center;">→ ←</td> <td></td> </tr> <tr> <td></td> <td colspan="2" style="text-align: center;">internal resource reservation was unsuccessful</td> </tr> <tr> <td style="text-align: right;">REL RLC</td> <td style="text-align: center;">← →</td> <td style="text-align: right;">CANCEL 200 OK CANCEL 487 Request terminated ACK</td> </tr> </table>		BICC IAM	SUT	SIP INVITE SIP_MESSAGE_VA		→ ←			internal resource reservation was unsuccessful		REL RLC	← →	CANCEL 200 OK CANCEL 487 Request terminated ACK
BICC IAM	SUT	SIP INVITE SIP_MESSAGE_VA												
	→ ←													
	internal resource reservation was unsuccessful													
REL RLC	← →	CANCEL 200 OK CANCEL 487 Request terminated ACK												

TP301053	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1.1															
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message																
SIP selection criteria:	Based on table 14																
ISUP selection criteria:																	
Test purpose:	<p>Ensure that the SUT in the Idle state on receipt of a IAM message, with the Transmission Medium Requirement (TMR) parameter set to TMR_VALUE:</p> <ul style="list-style-type: none"> • sends an INVITE message containing the media description defined with the "a = " "b = " and "m=" lines set to a_b_m_LINE_VALUE. 																
SIP Parameter values:	INVITE: a_b_m_LINE_VALUE																
ISUP Parameter values:	IAM: TMR: ISUP_TMR																
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP IAM</td> <td style="width: 50%; text-align: center;">SUT</td> <td style="width: 25%;">SIP INVITE</td> </tr> <tr> <td style="text-align: right;">ACM</td> <td style="text-align: center;">→ ←</td> <td style="text-align: right;">180 Ringing</td> </tr> <tr> <td style="text-align: right;">ANM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">200 OK INVITE</td> </tr> <tr> <td></td> <td colspan="2" style="text-align: center;">Conversation</td> </tr> <tr> <td style="text-align: right;">REL RLC</td> <td style="text-align: center;">→ ←</td> <td style="text-align: right;">BYE 200 OK BYE</td> </tr> </table>		ISUP IAM	SUT	SIP INVITE	ACM	→ ←	180 Ringing	ANM	←	200 OK INVITE		Conversation		REL RLC	→ ←	BYE 200 OK BYE
ISUP IAM	SUT	SIP INVITE															
ACM	→ ←	180 Ringing															
ANM	←	200 OK INVITE															
	Conversation																
REL RLC	→ ←	BYE 200 OK BYE															

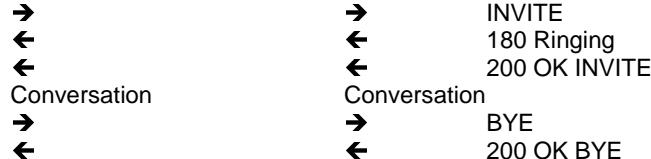
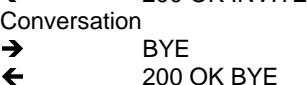
TP301054	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1.1																														
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message																															
SIP selection criteria:	Based on table 15																															
ISUP selection criteria:																																
Test purpose:	Ensure that the SUT in the Idle state on receipt of an IAM message, with the user information parameter set to USI_VALUE: <ul style="list-style-type: none"> sends an INVITE message, with the media description defined with the "a = " "b =" and "m=" lines set to a_b_m_LINE_VALUE. 																															
SIP Parameter values:	INVITE: a_b_m_LINE_VALUE																															
ISUP Parameter values:	IAM: TMR: ISUP_USI																															
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP</td> <td style="width: 25%; text-align: center;">→</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">→</td> <td style="width: 25%;">SIP</td> </tr> <tr> <td>IAM</td> <td>←</td> <td></td> <td>←</td> <td>INVITE</td> </tr> <tr> <td>ACM</td> <td>←</td> <td></td> <td>←</td> <td>180 Ringing</td> </tr> <tr> <td>ANM</td> <td>←</td> <td>Conversation</td> <td>←</td> <td>200 OK INVITE</td> </tr> <tr> <td>REL</td> <td>→</td> <td></td> <td>→</td> <td>BYE</td> </tr> <tr> <td>RLC</td> <td>←</td> <td></td> <td>←</td> <td>200 OK BYE</td> </tr> </table>	ISUP	→	SUT	→	SIP	IAM	←		←	INVITE	ACM	←		←	180 Ringing	ANM	←	Conversation	←	200 OK INVITE	REL	→		→	BYE	RLC	←		←	200 OK BYE	
ISUP	→	SUT	→	SIP																												
IAM	←		←	INVITE																												
ACM	←		←	180 Ringing																												
ANM	←	Conversation	←	200 OK INVITE																												
REL	→		→	BYE																												
RLC	←		←	200 OK BYE																												

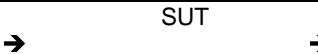
Table 14

Values for test purposes TP301053						
	ISUP	SDP - a_b_m_LINE_VALUE				
	TMR parameter	m= line		b= line	a= line	
	TMR codes	<media>	<transport>	<fmt-list>	<modifier>:<b andwidth- value>	rtpmap:<dynamic-PT> <encoding name>/<clock rate>[/encoding parameters>
VA_01	"speech"	Audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
VA_02	"speech"	Audio	RTP/AVP	Dynamic PT (and possibly a second Dynamic PT)	AS:64	rtpmap:<dynamic-PT> PCMU/8000 (and possibly rtpmap:<dynamic-PT> PCMA/8000)
VA_03	"speech"	Audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_04	"speech"	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> PCMA/8000
VA_05	"3,1 KHz audio"	Audio	RTP/AVP	0 and/or 8	AS:64	rtpmap:0 PCMU/8000 and/or rtpmap:8 PCMA/8000
VA_06	"3,1 KHz audio"	Audio	RTP/AVP	0 (and possibly 8)	AS:64	rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
VA_07	"3,1 KHz audio"	Audio	RTP/AVP	8	AS:64	rtpmap:8 PCMA/8000
VA_08	"64 kbit/s unrestricted"	Audio	RTP/AVP	9	AS:64	rtpmap:9 G722/8000
VA_09	"64 kbit/s unrestricted"	Audio	RTP/AVP	Dynamic PT	AS:64	rtpmap:<dynamic-PT> CLEARMODE/8000

Table 15

Values for test purposes TP301053, TP301054								
VA	ISUP			SDP - a_b_m_LINE_VALUE				
		USI parameter	HLC IE in ATP	m= line			b= line	a= line
	TMR	Information Transport Capability	User Information Layer 1 Protocol Indicator	High Layer Characteristics Identification	<media>	<transport>	<fmt-list>	<modifier>:<bandwidth-value>
VA_01	"speech"	"Speech"	"G.711 μ-law"	Ignore	audio	RTP/AVP	0 (and possibly 8) (see note 1)	AS:64
								rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000) See note 1
VA_02	"speech"	"Speech"	"G.711 μ-law"	Ignore	audio	RTP/AVP	Dynamic PT (and possibly a second Dynamic PT) (see note 1)	AS:64
								rtpmap:<dynamic-PT> PCMU/8000 (and possibly rtpmap:<dynamic-PT> PCMA/8000) (see note 1)
VA_03	"speech"	"Speech"	"G.711 A-law"	Ignore	audio	RTP/AVP	8	AS:64
VA_04	"speech"	"Speech"	"G.711 A-law"	Ignore	audio	RTP/AVP	Dynamic PT	AS:64
VA_05	"3,1 KHz audio"	USI Absent		Ignore	audio	RTP/AVP	0 and/or 8 (see note 1)	AS:64
VA_06	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 μ-law"		audio	RTP/AVP	0 (and possibly 8)	AS:64
								rtpmap:0 PCMU/8000 (and possibly rtpmap:8 PCMA/8000)
VA_07	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"		audio	RTP/AVP	8	AS:64
VA_08	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 μ-law"	"Facsimile Group 2/3"	image	udptl	t38	AS:64
VA_09	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	image	udptl	t38	AS:64
VA_10	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 μ-law"	"Facsimile Group 2/3"	image	tcptl	t38	AS:64
VA_11	"3,1 KHz audio"	"3,1 KHz audio"	"G.711 A-law"	"Facsimile Group 2/3"	image	tcptl	t38	AS:64
VA_12	"64 kbit/s unrestricted"	"Unrestricted digital inf. W/tone/ann."	N/A	Ignore	audio	RTP/AVP	9	AS:64
VA_13	"64 kbit/s unrestricted"	"Unrestricted digital information"	N/A	Ignore	Audio	RTP/AVP	Dynamic PT	AS:64
NOTE 1: <bandwidth value> for <modifier> of AS is evaluated to be B kbit/s.								
NOTE 2: CLEARMODE has been standardized.								

TP301055	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1.1
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in the Idle state on receipt of a IAM message, with the user information parameter set to USI_VALUE and Transmission Medium Requirement (TMR) parameter set to TMR_VALUE: <ul style="list-style-type: none"> • sends an INVITE message with the media description defined with the "a = " "b =" and "m=" lines set to a_b_m_LINE_VALUE; • ensure that the SUT is capable of encoding the SDP for the AMR codec, which is specified in RFC 3267 [12]: "RTP payload format and file storage format for the Adaptive Multi-Rate (AMR) and Adaptive Multi-Rate Wideband (AMR-WB) audio codec". 	
SIP Parameter values:	INVITE: a_b_m_LINE_VALUE	
ISUP Parameter values:	IAM: TMR: ISUP_USI	
Comments:	ISUP IAM ACM ANM REL RLC	SUT  Conversation  Conversation SIP INVITE 180 Ringing 200 OK INVITE BYE 200 OK BYE

TP301056	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1.2
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM: <ul style="list-style-type: none"> • to the addr-spec component of the To header field in the INVITE message. 	
SIP Parameter values:	INVITE: To: ...	
ISUP Parameter values:		
Comments:	ISUP IAM	SUT  SIP INVITE

TP301057	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1.2
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM: <ul style="list-style-type: none"> • to the addr-spec component of the To header field which shall include the "user=phone" URI parameter if the To header field contains a sip: URI. 	
SIP Parameter values:	INVITE: To: sip:; user=phone	
ISUP Parameter values:		
Comments:	ISUP IAM →	SUT → SIP INVITE

TP301058	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1.2
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter of the IAM and the followed SAM: <ul style="list-style-type: none"> • to the addr-spec component of the To header field. 	
SIP Parameter values:	INVITE: To:	
ISUP Parameter values:		
Comments:	ISUP IAM SAM SAM → → →	SUT → SIP INVITE

TP301059	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1.2
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT is mapping in the Called Party Number parameter contained in the Called Party address information of the IAM and following SAM: <ul style="list-style-type: none"> • to the addr-spec component of the To header field which shall include the "user=phone" URI parameter if the To header field contains a sip: URI. 	
SIP Parameter values:	INVITE: To: sip:; user=phone	
ISUP Parameter values:		
Comments:	ISUP IAM SAM SAM → → →	SUT → SIP INVITE

TP301060	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1.4
TSS reference:	ISUP-SIP/Basic call/ Sending of the Initial Address message (IAM)/	
SIP selection criteria:		
ISUP selection criteria:	PICS 4/3	
Test purpose:	Ensure that the SUT shall derive the Max-Forwards header field value from the Hop Counter parameter value by applying a factor. The Max-Forwards header field value for a given message should never increase and should decrease by at least 1 with each successive visit to an IWU, regardless of intervening interworking, and similarly for Max-Hop Counter in the BICC/ISUP domain.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	The initial and successively mapped values of Hop Counter should be large enough to accommodate the maximum number of hops that might be expected of a validly routed call. ISUP IAM → SUT → SIP INVITE	

TP301061	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1.2
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 1/9	
ISUP selection criteria:	PICS 1/8	
Test purpose:	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, Nature of address = "International number" of the IAM: <ul style="list-style-type: none">• to the addr-spec component of the To header field in the INVITE message;• the format of the To header field is "+CC+NDC+SN";• the forward address information is derived from the userinfo component of the INVITE Request-URI.	
SIP Parameter values:	INVITE: To: ...	
ISUP Parameter values:		
Comments:	ISUP IAM → SUT → SIP INVITE	

TP301062	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1.2
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 1/9	
ISUP selection criteria:	NOT PICS 1/8	
Test purpose:	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, Nature of address = "National (significant) number" of the IAM:	<ul style="list-style-type: none"> • to the addr-spec component of the To header field in the INVITE message; • the format of the To header field is "+CC+NDC+SN"; • the forward address information is derived from the userinfo component of the INVITE Request-URI.
SIP Parameter values:	INVITE: To: ...	
ISUP Parameter values:		
Comments:	ISUP IAM	SUT → SIP INVITE

TP301063	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1.2
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 1/9	
ISUP selection criteria:	PICS 1/8	
Test purpose:	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, Nature of address = "International number" of the IAM and the following SAM:	<ul style="list-style-type: none"> • to the addr-spec component of the To header field; • the format of the To header field is "+CC+NDC+SN"; • the forward address information is derived from the userinfo component of the INVITE Request-URI.
SIP Parameter values:	INVITE: To:	
ISUP Parameter values:		
Comments:	ISUP IAM SAM SAM	SUT → → → SIP INVITE

TP301064	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1.2
TSS reference:	ISUP-SIP/Basic call/ Sending of the INVITE message	
SIP selection criteria:	PICS 1/9	
ISUP selection criteria:	NOT PICS 1/8	
Test purpose:	Ensure that the SUT is mapping the Called Party address information contained in the Called Party Number parameter, Nature of address = "National (significant) number" of the IAM and the following SAM: <ul style="list-style-type: none"> • to the addr-spec component of the To header field; • The format of the To header field is "+CC+NDC+SN"; • the forward address information is derived from the userinfo component of the INVITE Request-URI. 	
SIP Parameter values:	INVITE: To:	
ISUP Parameter values:		
Comments:	ISUP IAM SAM SAM	SUT → → → SIP INVITE

6.2.2.2 Receipt of the SAM message after INVITE has been send

TP302001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after INVITE has been sent	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:		
Test purpose:	Ensure if the SUT is supporting en bloc addressing towards the SIP network, subsequent SAMs received after the SUT has sent the INVITE are ignored.	
SIP Parameter values:		
ISUP Parameter values:	SAM; subsequent number (PIXIT)	
Comments:	ISUP IAM SAM	SUT → → SIP INVITE

TP302002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1	
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent		
SIP selection criteria:	PICS 3/2		
ISUP selection criteria:	PICS 1/5		
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to indicate "continuity check not required":</p> <ul style="list-style-type: none"> • sends an INVITE message. <p>On receipt of a SAM from the ISUP the SUT shall:</p> <ol style="list-style-type: none"> 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP IAM SAM SAM	SUT → → →	SIP INVITE INVITE INVITE

TP302003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1	
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent		
SIP selection criteria:	PICS 3/2 AND NOT PICS 4/15		
ISUP selection criteria:	PICS 1/5 AND PICS 4/2		
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit".</p> <ul style="list-style-type: none"> • Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "continuity check successful". <p>On receipt of a SAM from the ISUP the SUT shall:</p> <ol style="list-style-type: none"> 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP IAM → SAM → COT → → INVITE SAM → → INVITE	SUT → → → → → →	SIP

TP302004	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1															
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent																
SIP selection criteria:	PICS 3/2 AND NOT PICS 4/15																
ISUP selection criteria:	PICS 1/5 AND PICS 4/2																
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check performed on previous circuit":</p> <ul style="list-style-type: none"> • Sends the INVITE after the receipt of the Continuity message with the Continuity Indicators parameter "continuity check successful". <p>On receipt of a SAM from the ISUP the SUT shall:</p> <ol style="list-style-type: none"> 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM. 																
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ISUP Parameter values:																	
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ISUP	SUT	SIP															
IAM	→																
SAM	→																
COT	→	→															
SAM	→	→															

TP302005	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	PICS 3/2 AND NOT PICS 4/15	
ISUP selection criteria:	PICS 1/5 AND PICS 4/2	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" sending of INVITE is delayed.</p> <p>INVITE message shall not be sent after the Continuity message was received with the Continuity Indicators parameter set to "continuity check failed".</p> <p>On receipt of a SAM from the ISUP the SUT shall:</p> <ol style="list-style-type: none"> 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM → SAM → COT →	SUT SIP

TP302007	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	PICS 3/2 AND NOT PICS 4/15	
ISUP selection criteria:	PICS 1/5 AND PICS 4/2	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set to "continuity check required on this circuit" sending of INVITE is delayed.</p> <p>INVITE shall not be sent after the ISUP timer T8 expires.</p> <p>On receipt of a SAM from the ISUP the SUT shall:</p> <ol style="list-style-type: none"> 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM → SAM → T8 expires REL ← RLC →	SUT SIP

TP302009	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1																		
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent																			
SIP selection criteria:	PICS 3/2 AND PICS 4/5 AND PICS 4/15																			
ISUP selection criteria:	PICS 1/5 AND PICS 4/2																			
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set "continuity check required on this circuit".</p> <ul style="list-style-type: none"> Sends an INVITE message after the reception of the Continuity message with the Continuity Indicators parameter set to "continuity check successful" and after the requested preconditions are met in the SIP network. <p>On receipt of a SAM from the ISUP the SUT shall:</p> <ol style="list-style-type: none"> Stop timer T_{oiw3} (if it is running). T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. All other contents of the new INVITE are interworked from the parameters of the original IAM. 																			
SIP Parameter values:																				
ISUP Parameter values:																				
Comments:	<p>The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied.</p> <table style="width: 100%; text-align: center;"> <tr> <td style="width: 33%;">ISUP</td> <td style="width: 33%;">SUT</td> <td style="width: 33%;">SIP</td> </tr> <tr> <td>IAM</td> <td>→</td> <td>INVITE</td> </tr> <tr> <td>SAM</td> <td>→</td> <td></td> </tr> <tr> <td>COT</td> <td>→</td> <td>183 Session Progress UPDATE</td> </tr> <tr> <td>SAM</td> <td>→</td> <td>200 OK UPDATE</td> </tr> <tr> <td></td> <td>→</td> <td>INVITE</td> </tr> </table>	ISUP	SUT	SIP	IAM	→	INVITE	SAM	→		COT	→	183 Session Progress UPDATE	SAM	→	200 OK UPDATE		→	INVITE	
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IAM	→	INVITE																		
SAM	→																			
COT	→	183 Session Progress UPDATE																		
SAM	→	200 OK UPDATE																		
	→	INVITE																		

TP302010	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1															
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent																
SIP selection criteria:	PICS 3/2 AND PICS 4/5 AND PICS 4/15																
ISUP selection criteria:	PICS 1/5 AND PICS 4/2																
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the Continuity Check indicator in the Nature of Connection Indicators parameter which is set or "continuity check performed on previous circuit":</p> <ul style="list-style-type: none"> Sends an INVITE message after the reception of the Continuity message with the Continuity Indicators parameter set to "continuity check successful" and after the requested preconditions are met in the SIP network. <p>On receipt of a SAM from the ISUP the SUT shall:</p> <ol style="list-style-type: none"> Stop timer T_{oiw3} (if it is running). T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. All other contents of the new INVITE are interworked from the parameters of the original IAM. 																
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ISUP Parameter values:																	
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ISUP	SUT	SIP															
IAM	→	INVITE															
SAM	→																
COT	→	183 Session Progress UPDATE															
SAM	→	200 OK UPDATE INVITE															

TP302011	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	PICS 3/2 AND NOT PICS 4/15	
ISUP selection criteria:	PICS 1/4 AND NOT PICS 4/2	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".</p> <p>The sending of the INVITE is delayed until all the following conditions are satisfied:</p> <ul style="list-style-type: none"> • Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received. • Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received. <p>On receipt of a SAM from the BICC the SUT shall:</p> <ol style="list-style-type: none"> 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	BICC IAM SAM COT SAM	<p style="text-align: center;">SUT</p> <p style="text-align: center;">→</p> <p style="text-align: right;">INVITE</p> <p style="text-align: right;">INVITE</p>

TP302012	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1										
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent											
SIP selection criteria:	PICS 3/2 AND NOT PICS 4/15											
ISUP selection criteria:	PICS 1/4 AND PICS 4/2											
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".</p> <p>The sending of the INVITE is delayed until all the following conditions are satisfied:</p> <ul style="list-style-type: none"> • Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received. • APM with Action indicator set to "Connected" - for the forward bearer set-up cases (with, or without bearer control tunnelling) where the incoming Connect Type is "notification required", and for the fast set-up (backward) case. <p>On receipt of a SAM from the BICC the SUT shall:</p> <ol style="list-style-type: none"> 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM. 											
SIP Parameter values:												
ISUP Parameter values:												
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SUT	SIP											
➔												
➔												
➔	➔											
➔	➔											

TP302013	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	PICS 3/2 AND NOT PICS 4/15	
ISUP selection criteria:	PICS 1/4 AND PICS 4/2	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".</p> <p>The sending of the INVITE delays until all the following conditions are satisfied:</p> <ul style="list-style-type: none"> Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received. Bearer Set-up Connect indication - for the backward bearer set-up case was received. <p>On receipt of a SAM from the BICC the SUT shall:</p> <ol style="list-style-type: none"> Stop timer T_{oiw3} (if it is running). T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. All other contents of the new INVITE are interworked from the parameters of the original IAM. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	BICC IAM SAM COT SAM	SUT → → → → → SIP INVITE INVITE

TP302014	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1	
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent		
SIP selection criteria:	PICS 3/2 AND NOT PICS 4/15		
ISUP selection criteria:	PICS 1/4 AND PICS 4/2		
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".</p> <p>The sending of the INVITE delays until all the following conditions are satisfied:</p> <ul style="list-style-type: none"> • Continuity message, with the Continuity Indicators parameter set to "continuity" shall be received. • BNC set-up success indication for cases using bearer control tunnelling was received. <p>On receipt of a SAM from the BICC the SUT shall:</p> <ol style="list-style-type: none"> 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	BICC IAM SAM COT SAM	<p style="text-align: center;">SUT</p> <p style="text-align: center;">→</p>	<p style="text-align: center;">SIP</p> <p style="text-align: center;">INVITE</p> <p style="text-align: center;">INVITE</p>

TP302015	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	PICS 3/2 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/4 AND PICS 4/2	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".</p> <p>Sends the INVITE message. The events</p> <ul style="list-style-type: none"> • Continuity message, with the Continuity Indicators parameter set to "continuity" was received; • Bearer Set-up indication - for the forward bearer set-up case where the incoming Connect Type is "notification not required" was received. <p>are indicating the successful completion of bearer set-up.</p> <p>On receipt of a SAM from the BICC the SUT shall:</p> <ol style="list-style-type: none"> 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied.	<pre> sequenceDiagram participant BICC participant SUT participant SIP BICC->>SUT: IAM SUT->>BICC: SAM BICC->>SUT: COT SUT->>BICC: SAM SUT->>SIP: INVITE SIP->>SUT: 183 Session Progress UPDATE SUT->>SIP: 200 OK UPDATE SIP->>SUT: INVITE </pre>

TP302016	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	PICS 3/2 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/4 AND PICS 4/2	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".</p> <p>Sends the INVITE message. The events:</p> <ul style="list-style-type: none"> • Continuity message, with the Continuity Indicators parameter set to "continuity" was received; • APM with Action indicator set to "Connected" - for the forward bearer set-up cases (with, or without bearer control tunnelling) where the incoming Connect Type is "notification required", and for the fast set-up (backward) case. <p>are indicating the successful completion of bearer set-up.</p> <p>On receipt of a SAM from the BICC the SUT shall:</p> <ol style="list-style-type: none"> 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied.	<pre> sequenceDiagram participant BICC participant SUT participant SIP BICC->>SUT: IAM BICC->>SUT: SAM SUT->>BICC: COT BICC->>SUT: SAM SUT->>SIP: 183 Session Progress UPDATE SUT->>SIP: 200 OK UPDATE SUT->>SIP: INVITE </pre>

TP302017	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	PICS 3/2 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/4 AND PICS 4/2	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message indicating "COT to be expected".</p> <p>Sends the INVITE message. The events:</p> <ul style="list-style-type: none"> • Continuity message, with the Continuity Indicators parameter set to "continuity" was received; • Bearer Set-up Connect indication - for the backward bearer set-up case was received. <p>are indicating the successful completion of bearer set-up.</p> <p>On receipt of a SAM from the BICC the SUT shall:</p> <ol style="list-style-type: none"> 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied.	<pre> sequenceDiagram participant BICC participant SUT participant SIP BICC->>SUT: IAM BICC->>SUT: SAM SUT->>BICC: COT BICC->>SUT: SAM SUT->>BICC: 183 Session Progress UPDATE SUT->>BICC: 200 OK UPDATE SUT->>BICC: INVITE </pre>

TP302018	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	PICS 3/2 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 1/4 AND PICS 4/2	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing indicating "COT to be expected".</p> <p>Sends the INVITE message. The events:</p> <ul style="list-style-type: none"> • Continuity message, with the Continuity Indicators parameter set to "continuity" was received; • BNC set-up success indication for cases using bearer control tunnelling was received. <p>are indicating the successful completion of bearer set-up.</p> <p>On receipt of a SAM from the BICC/ISUP the SUT shall:</p> <ol style="list-style-type: none"> 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: <ol style="list-style-type: none"> a) The Request-URI and the To header field of the new INVITE shall contain all digits received so far for this call. b) A new INVITE with the same Call-ID and From header (including tag) as the previous INVITE is sent. c) The new INVITE shall contain a new SDP offer. The O-IWU may re-use any resources that have already been reserved for this call. This re-use of existing reserved resources shall be reflected within the precondition attributes for the SDP parameters in question. d) All other contents of the new INVITE are interworked from the parameters of the original IAM. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	The O-IWU should initiate the precondition signalling procedure using the SDP Offer in the INVITE. The precondition signalling is concluded upon sending (within an SDP offer-answer exchange) the confirmation of a precondition being met. The SDP Offer or Answer carrying the confirmation of a precondition being met is sent when the conditions to send a INVITE message are satisfied.	<pre> sequenceDiagram participant BICC participant SUT participant SIP BICC->>SUT: IAM SUT->>BICC: SAM BICC->>SUT: COT SUT->>BICC: SAM SUT->>SIP: SIP INVITE SIP->>SUT: 183 Session Progress UPDATE SUT->>SIP: 200 OK UPDATE SIP->>SUT: INVITE </pre>

TP302019	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.2.1	
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent		
SIP selection criteria:	PICS 3/2		
ISUP selection criteria:	PICS 1/4		
Test purpose:	The SUT in Idle state, on receipt of an IAM message sends a INVITE message. On receipt of a SAM from the BICC/ISUP the SUT shall: 1) Stop timer T_{oiw3} (if it is running). 2) T_{oiw2} shall be restarted and the SUT shall invoke the following procedures: a) ensure that if timer T_{oiw2} has expired, subsequent SAMs received; b) after the SUT has sent the INVITE are ignored.		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP IAM SAM SAM	SUT → → → → T_{oiw2} expired	SIP → → → → INVITE INVITE

TP302020	SIP reference: RFC 3261	ISUP reference: Q.1912.5 § 7.2.1
TSS reference:	ISUP-SIP/Basic call/Receipt of SAM after invite has been sent	
SIP selection criteria:	PICS 3/2	
ISUP selection criteria:	PICS 3/8	
Test purpose:	The SUT in Idle state, on receipt of an IAM message On receipt of a SAM from the BICC/ISUP the SUT shall: • sends a INVITE message if the minimum number of digits for routing the call has been received in the IAM and the SAM T_{oiw1} and T_{oiw2} shall be started and the SUT shall invoke the following procedures: • ensure that if timer T_{oiw2} has expired, subsequent SAMs received after the SUT has sent the INVITE are ignored.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM SAM SAM	SUT → → → → T_{oiw2} expired → → → → INVITE

6.2.2.3 Sending of the ACM message

TP303001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) a) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/24	
ISUP selection criteria:	PICS 4/9	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication:</p> <ul style="list-style-type: none"> • Sends the INVITE message to called user. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP not used all the way ISDN access indicator: "terminating access non-ISDN"</p>	
Comments:	ISUP IAM ACM	SUT  SIP INVITE

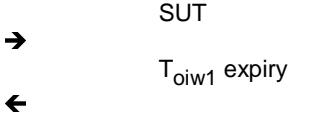
TP303002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) a) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND NOT PICS 1/9	
ISUP selection criteria:	PICS 4/9	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication:</p> <ul style="list-style-type: none"> • Sends the INVITE message to called user. • Sends the ACM message with the CPS indicator set to " no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)</p>	
Comments:	ISUP IAM ACM	SUT  SIP INVITE

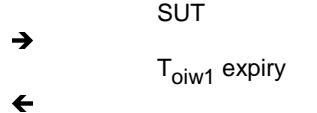
TP303003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) b) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1AND NOT PICS 4/24	
ISUP selection criteria:	PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan: <ul style="list-style-type: none"> • Sends the INVITE message to the called user. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP not used all the way ISDN access indicator: "terminating access non-ISDN"	
Comments:	ISUP IAM ACM	SUT → ← → SIP INVITE

TP303004	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) b) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1	
ISUP selection criteria:	PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan: <ul style="list-style-type: none"> • Sends the INVITE message to called user. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM ACM	SUT → ← → SIP INVITE

TP303005	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) c) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/24	
ISUP selection criteria:	PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party: <ul style="list-style-type: none"> • Sends the INVITE message to the called user. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : interworking encountered (1) ISUP indicator : ISUP not used all the way ISDN access indicator : "terminating access non-ISDN"	
Comments:	ISUP IAM ACM	SUT → ← → SIP INVITE

TP303006	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) c) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1	
ISUP selection criteria:	PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party: <ul style="list-style-type: none"> • Sends the INVITE message to called user. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM ACM	SUT → ← → SIP INVITE

TP303007	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) d), 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/24	
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{oiw1} after the receipt of the latest address message:</p> <ul style="list-style-type: none"> • Sends the INVITE message to the called user. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP not used all the way ISDN access indicator: "terminating access non-ISDN"</p>	
Comments:	<p>ISUP IAM  ACM</p>	<p>SUT  T_{oiw1} expiry INVITE</p>

TP303008	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) d), 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1	
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{oiw1} after the receipt of the latest address message:</p> <ul style="list-style-type: none"> • Sends the INVITE message to called user. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)</p>	
Comments:	<p>ISUP IAM  ACM</p>	<p>SUT  T_{oiw1} expiry INVITE</p>

TP303010	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1 and 7.3.1
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/2	
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the minimum number of digits required for routing the call has been received (start timer T_{oiw2} and invoke the appropriate outgoing SIP signalling procedure):</p> <ul style="list-style-type: none"> Sends an INVITE message to the called user and after the expiration of T_{oiw2}. Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM SAM SAM ACM	SUT → → → → T_{oiw2} expiry → ← INVITE

TP303011	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) a) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/24	
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication receipt of a 180 Ringing message: <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : subscriber free (01) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : interworking encountered (1) ISUP indicator : ISUP not used all the way ISDN access indicator : "terminating access non-ISDN"	
Comments:	ISUP IAM ACM	SUT → ← SIP INVITE 180 Ringing

TP303012	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) a) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1	
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication, on receipt of a 180 Ringing message:</p> <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: subscriber free (01) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)</p>	
Comments:	ISUP IAM ACM	SUT → ← SIP INVITE 180 Ringing

TP303013	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) b) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1	
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan on receipt of a 180 Ringing message:</p> <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: subscriber free (01) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP not used all the way ISDN access indicator: "terminating access non-ISDN"</p>	
Comments:	ISUP IAM ACM	SUT → ← SIP INVITE 180 Ringing

TP303014	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) b) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1	
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan on receipt of a 180 Ringing message: <ul style="list-style-type: none"> • Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : subscriber free (01) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM ACM	SUT → ← → ← SIP INVITE 180 Ringing

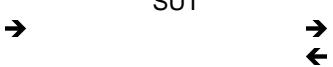
TP303015	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) c) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/24	
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party on receipt of a 180 Ringing message: <ul style="list-style-type: none"> • Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : subscriber free (01) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : interworking encountered (1) ISUP indicator : ISUP not used all the way ISDN access indicator : "terminating access non-ISDN"	
Comments:	ISUP IAM ACM	SUT → ← → ← SIP INVITE 180 Ringing

TP303016	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) c) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1	
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party on receipt of a 180 Ringing message:</p> <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: subscriber free (01) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)</p>	
Comments:	ISUP IAM ACM	SUT → INVITE ← 180 Ringing

TP303017	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1 1) d), 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/24	
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{oiw1} after the receipt of the latest address message:</p> <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP not used all the way ISDN access indicator: "terminating access non-ISDN"</p>	
Comments:	ISUP IAM ACM	SUT → T_{oiw1} expiry ← INVITE

TP303018	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) d), 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{oiw1} after the receipt of the latest address message: <ul style="list-style-type: none"> • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM ACM	→ SUT T_{oiw1} expiry ← → SIP INVITE

TP303019	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1 a) and 7.3.2
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication receipt of a 183 Session Progress: <ul style="list-style-type: none"> • Sends the INVITE message to called user. • No BICC/ISUP message is sent backward. 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number	
Comments:	ISUP IAM	→ SUT → SIP ← INVITE 183 Session Progress

TP303020	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1 b) and 7.3.2
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan on receipt of a 183 Session Progress:	<ul style="list-style-type: none"> • No BICC/ISUP message is sent backward.
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number: complete number	
Comments:	ISUP IAM	SUT 

TP303021	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1 c) and 7.3.2
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party on receipt of a 183 Session Progress:	<ul style="list-style-type: none"> • No BICC/ISUP message is sent backward.
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM	SUT 

TP303022	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1 and 7.3.2
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{oiw1} after the receipt of the latest address message on receipt of a 183 Session Progress: <ul style="list-style-type: none"> • No BICC/ISUP message is sent backward. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM ACM	SUT SIP INVITE T_{oiw1} expiry 183 Session Progress

TP303023	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15 AND NOT PICS 4/24	
ISUP selection criteria:	PICS 4/2 AND PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number , the sending complete indication, and the continuity check is performed (ISUP) or COT is expected (BICC): <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : interworking encountered (1) ISUP indicator : ISUP used all the way ISDN access indicator : "terminating access non-ISDN"	
Comments:	ISUP IAM COT ACM	SUT SIP INVITE 183 Session Progress UPDATE

TP303024	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 4/2 AND PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number , the sending complete indication and the continuity check is performed (ISUP) or COT is expected (BICC): <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM COT ACM	SUT → → → ← → → SIP INVITE 183 Session Progress UPDATE

TP303025	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1 and 7.4												
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message													
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15 AND NOT PICS 4/24													
ISUP selection criteria:	PICS 4/2 AND PICS 4/9													
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan and the continuity check is performed (ISUP) or COT is expected (BICC):</p> <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 													
SIP Parameter values:														
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access non-ISDN"</p>													
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: right;">SIP</td> </tr> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> <tr> <td>COT</td> <td style="text-align: center;">→</td> <td style="text-align: right;">183 Session Progress</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">UPDATE</td> </tr> </table>	ISUP	SUT	SIP	IAM	→	INVITE	COT	→	183 Session Progress	ACM	←	UPDATE	
ISUP	SUT	SIP												
IAM	→	INVITE												
COT	→	183 Session Progress												
ACM	←	UPDATE												

TP303026	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 4/2 AND PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan and the continuity check is performed (ISUP) or COT is expected (BICC), <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM COT ACM	SUT → → → ← → → SIP INVITE 183 Session Progress UPDATE

TP303027	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4	
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message		
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15 AND NOT PICS 4/24		
ISUP selection criteria:	PICS 4/2 AND PICS 4/9		
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party and the continuity check is performed (ISUP) or COT is expected (BICC):</p> <ul style="list-style-type: none"> Sends the INVITE message to called user. The SUT shall withhold sending ACM until a successful continuity indication has been received. Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 		
SIP Parameter values:			
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access non-ISDN"</p>		
Comments:	ISUP IAM → COT → ACM ←	SUT → ← →	SIP INVITE 183 Session Progress UPDATE

TP303028	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4	
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message		
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15		
ISUP selection criteria:	PICS 4/2 AND PICS 4/9		
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party and the continuity check is performed (ISUP) or COT is expected (BICC):</p> <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 		
SIP Parameter values:			
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)		
Comments:	ISUP IAM → COT → ACM ←	SUT → ← →	SIP INVITE 183 Session Progress UPDATE

TP303029	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4															
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message																
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15 AND NOT PICS 4/24																
ISUP selection criteria:	PICS 4/2																
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{oiw1} after the receipt of the latest address message and the continuity check is performed (ISUP) or COT is expected (BICC):</p> <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 																
SIP Parameter values:																	
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access non-ISDN"</p>																
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: right;">SIP</td> </tr> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> <tr> <td>COT</td> <td style="text-align: center;">→</td> <td style="text-align: right;">← 183 Session Progress</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">→ UPDATE</td> </tr> <tr> <td></td> <td style="text-align: center;">T_{oiw1} expiry</td> <td></td> </tr> </table>	ISUP	SUT	SIP	IAM	→	INVITE	COT	→	← 183 Session Progress	ACM	←	→ UPDATE		T_{oiw1} expiry		
ISUP	SUT	SIP															
IAM	→	INVITE															
COT	→	← 183 Session Progress															
ACM	←	→ UPDATE															
	T_{oiw1} expiry																

TP303030	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4												
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message													
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15													
ISUP selection criteria:	PICS 4/2													
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{oiw1} after the receipt of the latest address message and the continuity check is performed (ISUP) or COT is expected (BICC):</p> <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 													
SIP Parameter values:														
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)													
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: right;">SIP</td> </tr> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> <tr> <td>COT</td> <td style="text-align: center;">→</td> <td style="text-align: right;">← 183 Session Progress</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">→ UPDATE</td> </tr> </table> <p style="text-align: center;">T_{oiw1} expiry</p>	ISUP	SUT	SIP	IAM	→	INVITE	COT	→	← 183 Session Progress	ACM	←	→ UPDATE	
ISUP	SUT	SIP												
IAM	→	INVITE												
COT	→	← 183 Session Progress												
ACM	←	→ UPDATE												

TP303032	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4																												
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message																													
SIP selection criteria:	PICS 1/2 AND PICS 3/2 AND PICS 4/5 AND PICS 4/15																													
ISUP selection criteria:	PICS 4/2																													
Test purpose:	<p>Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the minimum number of digits required for routing the call has been received (start timer T_{oiw2} and invoke the appropriate outgoing SIP signalling procedure) and the continuity check is performed (ISUP) or COT is expected (BICC):</p> <ul style="list-style-type: none"> • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication (00)" or "ordinary subscriber (01)" or "payphone (10)". 																													
SIP Parameter values:																														
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)																													
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ISUP	→	SUT	SIP																											
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			183 Session Progress																											
COT	→		UPDATE																											
		T_{oiw2} expiry																												
ACM	←																													

TP303033	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15 AND NOT PICS 4/24	
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete , the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message: <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : "subscriber free (01)" Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : interworking encountered (1) ISUP indicator : ISUP used all the way ISDN access indicator : "terminating access non-ISDN"	
Comments:	ISUP IAM COT ACM	SUT → → → ← → → ← ← SIP INVITE 183 Session Progress UPDATE 180 RINGING

TP303034	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message: <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : subscriber free (01) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM COT ACM	SUT → → → ← → → ← ← SIP INVITE 183 Session Progress UPDATE 180 RINGING

TP303035	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1 and 7.4	
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message		
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15 AND NOT PICS 4/24		
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9		
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message:</p> <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 		
SIP Parameter values:			
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: subscriber free (01) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access non-ISDN"</p>		
Comments:	ISUP IAM → COT → ACM ←	SUT → → ←	SIP INVITE 183 Session Progress UPDATE 180 RINGING

TP303036	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message:</p> <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: subscriber free (01) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10)</p>	
Comments:	ISUP IAM → COT → ACM ←	SUT → INVITE ← 183 Session Progress → UPDATE ← 180 RINGING

TP303037	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1 and 7.3.1															
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message																
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15 AND NOT PICS 4/24																
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9																
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message:</p> <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 																
SIP Parameter values:																	
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: subscriber free (01) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access non-ISDN"</p>																
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ISUP	SUT	SIP															
IAM	→	INVITE															
	←	183 Session Progress															
COT	→	UPDATE															
ACM	←	180 RINGING															

TP303038	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1 and 7.3.1															
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message																
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND PICS 4/5 AND PICS 4/15																
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9																
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message:</p> <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 																
SIP Parameter values:																	
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Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">SIP</td> </tr> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: center;">INVITE</td> </tr> <tr> <td></td> <td style="text-align: center;">←</td> <td style="text-align: center;">183 Session Progress</td> </tr> <tr> <td>COT</td> <td style="text-align: center;">→</td> <td style="text-align: center;">UPDATE</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: center;">180 RINGING</td> </tr> </table>	ISUP	SUT	SIP	IAM	→	INVITE		←	183 Session Progress	COT	→	UPDATE	ACM	←	180 RINGING	
ISUP	SUT	SIP															
IAM	→	INVITE															
	←	183 Session Progress															
COT	→	UPDATE															
ACM	←	180 RINGING															

TP303039	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/15 AND NOT PICS 4/24	
ISUP selection criteria:	PICS 4/2 AND PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number , the sending complete indication, and the continuity check is performed (ISUP) or COT is expected (BICC): <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : interworking encountered (1) ISUP indicator : ISUP used all the way ISDN access indicator : "terminating access non-ISDN"	
Comments:	ISUP IAM COT ACM	SUT → → ← SIP INVITE

TP303040	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND NOT PICS 4/15	
ISUP selection criteria:	PICS 4/2 AND PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number , the sending complete indication and the continuity check is performed (ISUP) or COT is expected (BICC): <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM COT ACM	SUT → → ← SIP INVITE

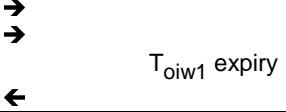
TP303041	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/15 AND NOT PICS 4/24	
ISUP selection criteria:	PICS 4/2 AND PICS 4/9	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan and the continuity check is performed (ISUP) or COT is expected (BICC):</p> <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access non-ISDN"</p>	
Comments:	ISUP IAM COT ACM <div style="text-align: center; margin-top: 10px;"> <pre> graph LR SUT[] --> ISUP SIP[SIP] SIP --> IAM SUT SUT --> COT SIP SIP --> ACM SUT SUT --> INVITE SIP </pre> </div>	SUT SIP INVITE

TP303042	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND NOT PICS 4/15	
ISUP selection criteria:	PICS 4/2 AND PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan and the continuity check is performed (ISUP) or COT is expected (BICC): <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM COT ACM	SUT → → ← SIP INVITE

TP303043	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/15 AND NOT PICS 4/24	
ISUP selection criteria:	PICS 4/2 AND PICS 4/9	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party and the continuity check is performed (ISUP) or COT is expected (BICC):</p> <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : interworking encountered (1) ISUP indicator : ISUP used all the way ISDN access indicator : "terminating access non-ISDN"	
Comments:	ISUP IAM COT ACM	SUT → → ← → INVITE

TP303044	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND NOT PICS 4/15	
ISUP selection criteria:	PICS 4/2 AND PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party and the continuity check is performed (ISUP) or COT is expected (BICC): <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM COT ACM	SUT → → ← SIP INVITE

TP303045	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/15 AND NOT PICS 4/24	
ISUP selection criteria:	PICS 4/2	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{oiw1} after the receipt of the latest address message and the continuity check is performed (ISUP) or COT is expected (BICC):</p> <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	<p>IAM; Called party number: complete number ACM, CPS indicator: no indication (00) Called party's category indicator: no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator: interworking encountered (1) ISUP indicator: ISUP used all the way ISDN access indicator: "terminating access non-ISDN"</p>	
Comments:	ISUP IAM COT ACM	SUT SIP INVITE

TP303046	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND NOT PICS 4/15	
ISUP selection criteria:	PICS 4/2	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by the expiration timer T_{oiw1} after the receipt of the latest address message and the continuity check is performed (ISUP) or COT is expected (BICC):</p> <ul style="list-style-type: none"> • Sends the INVITE message to called user. • The SUT shall withhold sending ACM until a successful continuity indication has been received. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM COT ACM	SUT  INVITE

TP303047	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4	
TSS reference:	ISUP-SIP /Basic call/Sending of the INVITE message		
SIP selection criteria:	PICS 1/2 AND PICS 3/2 AND NOT PICS 4/15		
ISUP selection criteria:	PICS 3/8 AND PICS 4/2		
Test purpose:	<p>Ensure that the SUT if overlap addressing is to be used toward the SIP network, on receipt of an IAM message containing the minimum number of digits required for routing the call has been received (start timer T_{oiw2} and invoke the appropriate outgoing SIP signalling procedure) and the continuity check is performed (ISUP) or COT is expected (BICC):</p> <ul style="list-style-type: none"> The SUT shall withhold sending ACM until a successful continuity indication has been received. Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 		
SIP Parameter values:			
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)		
Comments:	ISUP IAM SAM COT ACM	SUT → → → → T _{oiw2} expiry ←	SIP INVITE

TP303048	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4	
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message		
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/15 AND NOT PICS 4/24		
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9		
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message:</p> <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 		
SIP Parameter values:			
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : subscriber free (01) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : interworking encountered (1) ISUP indicator : ISUP used all the way ISDN access indicator : "terminating access non-ISDN"		
Comments:	ISUP IAM COT ACM	SUT → → → ←	SIP INVITE

TP303049	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND NOT PICS 4/15	
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message: <ul style="list-style-type: none"> • Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : subscriber free (01) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM COT ACM	SUT → → ← → SIP INVITE

TP303050	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/15 AND NOT PICS 4/24	
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message: <ul style="list-style-type: none"> • Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : subscriber free (01) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : interworking encountered (1) ISUP indicator : ISUP used all the way ISDN access indicator : "terminating access non-ISDN"	
Comments:	ISUP IAM COT ACM	SUT → → ← → SIP INVITE

TP303051	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 7.3.1 and 7.4
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND NOT PICS 4/15	
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the maximum number of digits used in the national numbering plan, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message: <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : subscriber free (01) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM COT ACM	SUT → → ← SIP → INVITE

TP303052	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.1 and 7.3.1
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/1 AND PICS 3/1 AND NOT PICS 4/15 AND NOT PICS 4/24	
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message: <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : subscriber free (01) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : interworking encountered (1) ISUP indicator : ISUP used all the way ISDN access indicator : "terminating access non-ISDN"	
Comments:	ISUP IAM COT ACM	SUT → → ← SIP → INVITE

TP303053	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1 and 7.3.1
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND NOT PICS 4/15	
ISUP selection criteria:	PICS 4/2 AND NOT PICS 4/9	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number where the end of address signalling is determined by analysis of the called party number to indicate that a sufficient number of digits has been received to route the call to the called party, the continuity check is performed (ISUP) or COT is expected (BICC) indication receipt of a 180 Ringing message:</p> <ul style="list-style-type: none"> Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : subscriber free (01) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10)	
Comments:	ISUP IAM COT ACM	SUT → → ← SIP INVITE

TP303054	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1 a) and 7.3.1
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 1/9 AND PICS 3/1	
ISUP selection criteria:	PICS 4/9	
Test purpose:	<p>Ensure that the SUT in Idle state, on receipt of an IAM message containing the complete called party number and the sending complete indication:</p> <ul style="list-style-type: none"> Sends the INVITE message to called user. Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "interworking encountered (1)", the ISUP indicator set to "ISUP not used all the way", the ISDN access indicator set to "terminating access non-ISDN". 	
SIP Parameter values:		
ISUP Parameter values:	IAM; Called party number : complete number ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : "interworking encountered (1)" ISUP indicator : "ISUP not used all the way" ISDN access indicator : "terminating access non-ISDN"	
Comments:	ISUP IAM ACM	SUT → ← SIP INVITE

TP303055	SIP reference: RFC 3261 [6]	ISUP reference: ETSI EN 383 001 § 7.3.1.1																								
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message																									
SIP selection criteria:	PICS 1/2 AND PICS 1/9 AND PICS 3/2 AND PICS 4/24																									
ISUP selection criteria:	NOT PICS 4/9																									
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message after the expiry of Toiw2: <ul style="list-style-type: none"> • Sends the INVITE message to called user. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "no interworking encountered (0)", the ISUP indicator set to "ISDN user part/BICC used all the way (0)", the ISDN access indicator set to "terminating access ISDN ".(1) 																									
SIP Parameter values:																										
ISUP Parameter values:	ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : no interworking encountered (0) ISUP indicator : ISDN user part/BICC used all the way (0) ISDN access indicator : terminating access ISDN (1)																									
	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">ISUP</th> <th style="text-align: center; width: 10%;">SUT</th> <th style="text-align: right; width: 60%;">SIP</th> </tr> </thead> <tbody> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ INVITE ← 404/484 → ACK</td> </tr> <tr> <td>SAM</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ INVITE ← 404/484 → ACK</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: right; vertical-align: bottom;">Toiw2 expiry</td> </tr> <tr> <td>CPG(alerting)</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← 180 Ringing</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← 200 OK INVITE → ACK</td> </tr> <tr> <td>REL</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ BYE</td> </tr> <tr> <td>RLC</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← 200 OK BYE</td> </tr> </tbody> </table>	ISUP	SUT	SIP	IAM	→	→ INVITE ← 404/484 → ACK	SAM	→	→ INVITE ← 404/484 → ACK	ACM	←	Toiw2 expiry	CPG(alerting)	←	← 180 Ringing	ANM	←	← 200 OK INVITE → ACK	REL	→	→ BYE	RLC	←	← 200 OK BYE	
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ANM	←	← 200 OK INVITE → ACK																								
REL	→	→ BYE																								
RLC	←	← 200 OK BYE																								

TP303056	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.3.1.1
TSS reference:	ISUP-SIP /Basic call/Sending of the ACM message	
SIP selection criteria:	PICS 1/2 AND PICS 1/9 AND PICS 3/1 AND PICS 4/24	
ISUP selection criteria:	NOT PICS 4/9	
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message, after the expiry of Toiw1: <ul style="list-style-type: none"> • Sends the INVITE message to called user. • Sends the ACM message with the CPS indicator set to "no indication (00)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "no interworking encountered (0)", the ISUP indicator set to "ISDN user part/BICC used all the way (0)", the ISDN access indicator set to "terminating access ISDN ".(1) 	
SIP Parameter values:		
ISUP Parameter values:	ACM, CPS indicator : no indication (00) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : no interworking encountered (0) ISUP indicator : ISDN user part/BICC used all the way (0) ISDN access indicator : terminating access ISDN (1)	
	ISUP IAM → SAM → SAM → ACM ← CPG(alerting) ← ANM ← REL → RLC ←	SUT Toiw1 expiry → INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK BYE

TP303057	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1, 1) c) and 7.3.1	
TSS reference:	ISUP-SIP /Basic call/ Sending of the ACM message		
SIP selection criteria:	PICS 1/2 AND PICS 3/1 AND PICS 4/24		
ISUP selection criteria:	NOT PICS 4/9		
Test purpose:	Ensure that the SUT in Idle state, on receipt of an IAM message the SUT sends out an INVITE, on receipt of a 180 Ringing message: <ul style="list-style-type: none"> • Sends the ACM message with the CPS indicator set to "subscriber free (01)", the Called party's category indicator set to "no indication(00)" or "ordinary subscriber (01)" or "payphone (10)", the interworking indicator set to "no interworking encountered" (0), the ISUP indicator set to " ISDN user part/BICC used all the way (0)", the ISDN access indicator set to "terminating access ISDN".(1). 		
SIP Parameter values:			
ISUP Parameter values:	ACM, CPS indicator : subscriber free (01) Called party's category indicator : no indication(00) or ordinary subscriber (01) or payphone (10) interworking indicator : no interworking encountered (0) ISUP indicator : ISDN user part/BICC used all the way (0) ISDN access indicator : terminating access ISDN (1)		
	ISUP IAM → ACM ← ANM ← REL → RLC ←	SUT → INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK BYE	SIP → INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK BYE

6.2.2.4 Sending of the CPG message

TP304001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1 and 7.3.1	
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message		
SIP selection criteria:	PICS 3/1		
ISUP selection criteria:			
Test purpose:	Ensure that the SUT, having sent a ACM message with called party status "no indication" on receipt of a 180 Ringing message: <ul style="list-style-type: none"> • Sends the CPG message with the with the event indicator set to "Alerting". 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP IAM → ACM ← CPG ←	SUT → INVITE T _{oiw1} expiry ← 180 Ringing	SIP → INVITE ← 180 Ringing

TP304002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.1 and 7.3.1
TSS reference:	ISUP-SIP /Basic call/ Sending of the CPG message	
SIP selection criteria:	PICS 3/1	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT, having sent a ACM message with called party status "no indication" on receipt of a 183 Session progress message: <ul style="list-style-type: none"> • No BICC/ISUP message is sent backward. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM ACM	SUT → T _{oiw1} expiry ← SIP INVITE ← 183 Session progress

6.2.2.5 Sending of the ANM message

TP305001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.5
TSS reference:	ISUP-SIP/Basic call/ Sending of the Answer Message (ANM)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT having sent the ACM message, on receipt of a 200 OK INVITE for this call, it shall stop timer T _{oiw2} (if running): <ul style="list-style-type: none"> • Send ANM as determined by BICC/ISUP procedures. • Stop any existing awaiting answer indication (e.g. ringing tone). 	
SIP Parameter values:	200 OK INVITE;	
ISUP Parameter values:	ANM;	
Comments:	ISUP IAM ACM ANM	SUT → 180 Ringing ← SIP INVITE → 200 OK INVITE ←

6.2.2.6 Sending of the CON message

TP306001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.5 and 7.5.1
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/	
SIP selection criteria:	PICS 1/1	
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT, having not sent the ACM message, on receipt of a 200 OK INVITE for this call, it shall stop timer T_{oiw2} (if running):</p> <ul style="list-style-type: none"> Send CON as determined by BICC/ISUP procedures. <p>Stop any existing awaiting answer indication (e.g. ringing tone) BCI encoded as follows:</p> <p style="padding-left: 20px;">Interworking indicator: interworking encountered</p> <p style="padding-left: 20px;">ISUP indicator: ISUP not used all the way</p> <p style="padding-left: 20px;">ISDN access indicator: terminating access non-ISDN</p> <p style="padding-left: 20px;">CPS indicator: no indication</p>	
SIP Parameter values:	200 OK INVITE;	
ISUP Parameter values:	CON; Interworking indicator: interworking encountered ISUP indicator: ISUP not used all the way ISDN access indicator: terminating access non-ISDN CPS indicator: no indication	
Comments:	ISUP IAM CON	SUT → ← SIP INVITE 200 OK INVITE

TP306002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.5 and 7.5.1
TSS reference:	ISUP-SIP/Basic call/ Sending of the Connect Message (CON)/	
SIP selection criteria:	PICS 1/2	
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT, having not sent the ACM message, on receipt of a 200 OK INVITE for this call, it shall stop timer T_{oiw2} (if running):</p> <ul style="list-style-type: none"> Send CON as determined by BICC/ISUP procedures. <p>Stop any existing awaiting answer indication (e.g. ringing tone) BCI encoded as follows:</p> <p style="padding-left: 40px;">interworking indicator: INT_IND_VAL (PIXIT)</p> <p style="padding-left: 40px;">ISUP indicator: ISUP_IND_ID (PIXIT)</p> <p style="padding-left: 40px;">ISDN access indicator: ISDN_ACC_IND_VAL (PIXIT)</p> <p style="padding-left: 40px;">CPS indicator: no indication</p>	
SIP Parameter values:	200 OK INVITE;	
ISUP Parameter values:	CON; interworking indicator: INT_IND_VAL (PIXIT) ISUP indicator: ISUP_IND_ID (PIXIT) ISDN access indicator: ISDN_ACC_IND_VAL (PIXIT) CPS indicator: no indication	
Comments:	ISUP IAM CON	SUT → ← → ← SIP INVITE 200 OK INVITE

6.2.2.7 Receipt of the Release message (REL)

TP307001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.1, 1)
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM but before an INVITE has been sent. On receipt of a REL message:	<ul style="list-style-type: none"> no action is required on the SIP side other than to terminate local procedures if any are in progress.
SIP Parameter values:		
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT) location: LOC_ISUP (PIXIT)	
Comments:	ISUP IAM REL RLC	SUT → → ← SIP

TP307002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.1, 2)
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message before a 200 OK (any) response message has been received which establishes a confirmed dialogue:	<ul style="list-style-type: none"> • The SUT shall hold the REL message until a SIP 200 OK INVITE response has been received. • The SUT shall send a BYE request.
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM REL RLC	<p style="text-align: center;">SUT</p> <pre> graph LR SUT[] --> IAM SIP[SIP] SIP --> INVITE SUT SUT --> REL SIP SIP --> 200 OK INVITE SUT SUT --> ACK SIP SIP --> ACK SUT SUT --> BYE SIP SIP --> 200 OK BYE SUT </pre>

TP307003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.1, 2), 3)
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message before a 200 OK SIP response message has been received:	<ul style="list-style-type: none"> • The SUT shall hold the REL message. A CANCEL is sent when any SIP response was been received. • On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent.
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM REL RLC	<p style="text-align: center;">SUT</p> <pre> graph LR SUT[] --> IAM SIP[SIP] SIP --> INVITE SUT SUT --> REL SIP SIP --> 100 TRYING SUT SUT --> CANCEL SIP SIP --> 200 OK INVITE SUT SUT --> ACK SIP SIP --> ACK SUT SUT --> BYE SIP SIP --> 200 OK CANCEL SUT SUT --> BYE SIP SIP --> 200 OK BYE SUT </pre>

TP307004	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.1, 2), 3)																																													
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/																																														
SIP selection criteria:	NOT PICS 4/10																																														
ISUP selection criteria:																																															
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message before an early dialogue with the message defined as SIP_MESSAGE_VA has been established:</p> <ul style="list-style-type: none"> The SUT shall hold the REL message until a SIP_MESSAGE_VA response has been received. The SUT shall send a CANCEL or BYE request. 																																														
SIP Parameter values:																																															
ISUP Parameter values:																																															
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TP307005	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.1, 4)																								
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/																									
SIP selection criteria:																										
ISUP selection criteria:	NOT PICS 4/10																									
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message after a 200 OK response message has been received:</p> <ul style="list-style-type: none"> The SUT shall hold the REL message until an ACK has been sent. The SUT shall send a BYE request. 																									
SIP Parameter values:																										
ISUP Parameter values:																										
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 20%;">ISUP</td> <td style="width: 20%; text-align: center;">SUT</td> <td style="width: 20%; text-align: center;">SIP</td> </tr> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td>INVITE</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td>180 Ringing</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> <td>200 OK INVITE</td> </tr> <tr> <td>REL</td> <td style="text-align: center;">→</td> <td>ACK</td> </tr> <tr> <td>RLC</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td></td> <td style="text-align: center;">→</td> <td>BYE</td> </tr> <tr> <td></td> <td style="text-align: center;">←</td> <td>200 OK BYE</td> </tr> </table>	ISUP	SUT	SIP	IAM	→	INVITE	ACM	←	180 Ringing	ANM	←	200 OK INVITE	REL	→	ACK	RLC	←			→	BYE		←	200 OK BYE	
ISUP	SUT	SIP																								
IAM	→	INVITE																								
ACM	←	180 Ringing																								
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REL	→	ACK																								
RLC	←																									
	→	BYE																								
	←	200 OK BYE																								

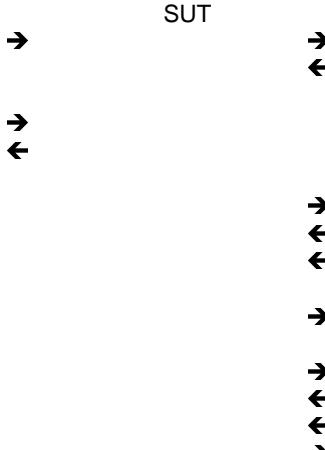
TP307006	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.1, 3)
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established:	<ul style="list-style-type: none"> • The SUT shall send a CANCEL or BYE request.
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM REL RLC CASE A terminated CASE B	SUT  SIP INVITE SIP_MESSAGE_VA CANCEL 200 OK CANCEL 487 Request ACK BYE 200 OK BYE 487 Request terminated ACK

Table 16

Values for test purposes TP307004; TP307006	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress

TP307007	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.1, 2), 4)	
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/		
SIP selection criteria:	PICS 4/10		
ISUP selection criteria:			
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message on receipt REL message before a 200 OK response (any) message has been received which establishes a confirmed dialogue:</p> <ul style="list-style-type: none"> The SUT shall hold the REL message until a SIP 200 OK INVITE response has been received. The SUT shall send a BYE request. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP. 		
SIP Parameter values:	cause value: CV_SIP (PIXIT)		
ISUP Parameter values:	<p>REL: cause value: CV_ISUP (PIXIT) location: LOC_ISUP (PIXIT)</p>		
Comments:	ISUP IAM REL RLC	SUT → → ←	SIP INVITE 200 OK INVITE ACK BYE 200 OK BYE

TP307008	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.1, 2), 3																														
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/																															
SIP selection criteria:	PICS 4/10																															
ISUP selection criteria:																																
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message before a 200 OK response message has been received:</p> <ul style="list-style-type: none"> The SUT shall hold the REL message. A CANCEL is sent when any SIP response was been received. On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP. 																															
SIP Parameter values:	BYE: cause value : CV_SIP (PIXIT)																															
ISUP Parameter values:	REL: cause value : CV_ISUP (PIXIT) location : LOC_ISUP (PIXIT)																															
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: right;">SIP</td> </tr> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> <tr> <td></td> <td style="text-align: center;">←</td> <td style="text-align: right;">100 TRYING</td> </tr> <tr> <td>REL</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>RLC</td> <td style="text-align: center;">←</td> <td></td> </tr> </table>	ISUP	SUT	SIP	IAM	→	INVITE		←	100 TRYING	REL	→		RLC	←		<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: right;">SIP</td> </tr> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> <tr> <td></td> <td style="text-align: center;">←</td> <td style="text-align: right;">100 TRYING</td> </tr> <tr> <td>REL</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>RLC</td> <td style="text-align: center;">←</td> <td></td> </tr> </table>	ISUP	SUT	SIP	IAM	→	INVITE		←	100 TRYING	REL	→		RLC	←	
ISUP	SUT	SIP																														
IAM	→	INVITE																														
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IAM	→	INVITE																														
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REL	→																															
RLC	←																															

TP307009	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.1, 3)
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message before an early dialogue with the message defined as SIP_MESSAGE has been established: <ul style="list-style-type: none"> • The SUT shall hold the REL message until a SIP_MESSAGE_VA response has been received. • The SUT shall send a CANCEL request or a BYE request. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP. 	
SIP Parameter values:	CANCEL: cause value: CV_SIP (PIXIT)	
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT) location: LOC_ISUP (PIXIT)	
Comments:	ISUP IAM → REL → RLC ← CASE A terminated CASE B terminated	SUT → SIP INVITE ← SIP_MESSAGE_VA → CANCEL ← 200 OK CANCEL ← 487 Request → ACK → BYE ← 200 OK BYE ← 487 Request → ACK

TP307010	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.1, 3)
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message after a 200 OK response message has been received: <ul style="list-style-type: none"> The SUT shall send a BYE request after the ACK has been sent. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP. 	
SIP Parameter values:	BYE: cause value: CV_SIP (PIXIT)	
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT) location: LOC_ISUP (PIXIT)	
Comments:	ISUP IAM → ACM ← ANM ← REL → RLC ←	SUT → SIP INVITE 180 Ringing 200 OK INVITE ACK → BYE ← 200 OK BYE

TP307011	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.1
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM with the complete called party number, sending an INVITE message. On receipt of a REL message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established: <ul style="list-style-type: none"> The SUT shall send a CANCEL or BYE request. The cause Value Indicator parameter defined as CV_ISUP shall be mapped to the Reason header field defined as CV_SIP. 	
SIP Parameter values:	CANCEL: cause value: CV_SIP (PIXIT)	
ISUP Parameter values:	REL: cause value: CV_ISUP (PIXIT) location: LOC_ISUP (PIXIT)	
Comments:	ISUP IAM → REL → RLC ← Case A terminated Case B terminated	SUT → SIP INVITE SIP_MESSAGE_VA → CASE A → 200 OK CANCEL → 487 Request → ACK → BYE → 200 OK BYE → 487 Request → ACK

Table 17

Values for test purpose TP307009; TP307011	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress

Table 18

Values for test purposes 307007 - 307011		
←SIP Message Reason header field CV_SIP		← REL Cause Indicators parameter CV_ISUP
VA_1	Normal call clearing # 16	Normal call clearing # 16
VA_2	Normal, unspecified # 31	Normal, unspecified # 31
VA_3	Temporary failure # 41	Temporary failure # 41
VA_4	Invalid message, unspecified # 95	Invalid message, unspecified # 95
VA_5	Recovery on timer expiry # 102	Recovery on timer expiry # 102
VA_6	Protocol error, unspecified # 111	Protocol error, unspecified # 111

Table 19: Mapping of Cause Indicators parameter into SIP Reason header fields

Cause indications parameter field	Value of parameter field	component of SIP Reason header field	Component value
-	-	Protocol	"ITU-T Rec Q.850 [5]"
Cause Value	"XX" (see note 1)	Protocol-cause	"cause= XX" (see note 1)
-	-	Reason-text	Should be filled with the definition text as stated in ITU-T Rec Q.850 [5] (see note 2)

NOTE 1: "XX" is the Cause Value as defined in ITU-T Rec Q.850 [5].

NOTE 2: Due to the fact that the Cause Indications parameter does not include the definition text as defined in Table1/ITU-T Rec Q.850 [5] this is based on provisioning in the O-IWU.

6.2.2.8 Sending of a REL message (REL) / receipt of a backward BYE

TP308001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.2
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/11	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM sends out an INVITE message and on receipt of a BYE message where a Reason header field with ITU-T Rec Q.850 [5] Cause Value is not included: <ul style="list-style-type: none"> sends a REL message with the Cause value Value No. 16 ("normal clearing"). 	
SIP Parameter values:		
ISUP Parameter values:	REL; Cause value "Normal call clearing"	
Comments:	ISUP IAM → SUT ACM ← → SIP ANM ← → INVITE REL ← → 180 Ringing RLC → → 200 OK INVITE Conversation	SIP INVITE 180 Ringing 200 OK INVITE BYE 200 OK BYE

TP308004	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.2
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	PICS 4/11	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM sends out a INVITE message and on receipt of a BYE message where a Reason header field with ITU-T Rec Q.850 [5] Cause Value is included: <ul style="list-style-type: none"> sends a REL message. The Cause Value is in the Reason header filed mapped to the ISUP Cause Value field in the ISUP REL. 	
SIP Parameter values:	BYE cause value: CV_SIP (PIXIT)	
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)	
Comments:	ISUP IAM → SUT ACM ← → SIP ANM ← → INVITE REL ← → 180 Ringing RLC → → 200 OK INVITE Conversation	SIP INVITE 180 Ringing 200 OK INVITE BYE 200 OK BYE

Table 20: Mapping of SIP Reason header fields into Cause Indicators parameter

component of SIP Reason header field	Component value	BICC/ISUP Parameter / field	value
Protocol	"ITU-T Rec Q.850 [5]"	Cause Indication parameter	-
protocol-cause	"cause = XX" (see note)	Cause Value	"XX" (see note)
-	-	Location	"network beyond interworking point"

NOTE: "XX" is the Cause Value as defined in ITU-T Rec Q.850 [5].

TP308007	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.6
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/11	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with ITU-T Rec Q.850 [5] Cause Value is not included defined as SIP_Failure_VA: <ul style="list-style-type: none"> • sends a REL message with the Cause value set to CV_ISUP. 	
SIP Parameter values:		
ISUP Parameter values:	REL; cause value: CV_ISUP	
Comments:	ISUP IAM REL RLC	SUT → ← → → SIP INVITE SIP_Failure_VA ACK

Table 21

Values for test purpose TP308007.		
VA	←REL (Cause Value) CV_ISUP	←4XX/5XX/6XX SIP message SIP_Failure_VA
VA_01	127 Interworking	400 Bad Request
VA_02	127 Interworking	402 Payment Required
VA_03	127 Interworking	403 Forbidden
VA_04	1 Unallocated number	404 Not Found
VA_05	127 Interworking	405 Method Not Allowed
VA_06	127 Interworking	406 Not Acceptable
VA_07	127 Interworking	408 Request Timeout
VA_08	22 Number changed (without diagnostic)	410 Gone
VA_09	127 Interworking	423 Interval Too Brief
VA_10	20 Subscriber absent	480 Temporarily Unavailable
VA_11	127 Interworking	481 Call/Transaction does not exist
VA_12	127 Interworking	482 Loop Detected
VA_13	127 Interworking	483 Too many hops
VA_14	127 Interworking	485 Ambiguous
VA_15	17 User busy	486 Busy Here
VA_16	127 Interworking	488 Not acceptable here
VA_17	127 Interworking	493 Undecipherable
VA_18	127 Interworking	500 Server Internal error
VA_19	127 Interworking	501 Not implemented
VA_20	127 Interworking	502 Bad Gateway
VA_21	127 Interworking	504 Server timeout
VA_22	17 User busy	600 Busy Everywhere
VA_23	21 Call rejected	603 Decline
VA_24	1 Unallocated number	604 Does not exist anywhere
VA_25	127 Interworking	606 Not acceptable

TP308008	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.6
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/12	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if the SIP Failure response is interworked to ISUP after receiving an IAM message sends out an INVITE message. On receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with ITU-T Rec Q.850 [5] Cause Value is not included defined as SIP_Failure_VA: <ul style="list-style-type: none"> • sends a REL message with the Cause value set to CV_ISUP. 	
SIP Parameter values:		
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)	
Comments:	ISUP IAM REL RLC	SUT → ← → → SIP INVITE SIP_Failure_VA ACK

Table 22

Values for test purposes TP308008		
VA	←REL (Cause Value) CV_ISUP	←4XX/5XX/6XX SIP message SIP_Failure_VA
VA_01	127 Interworking	401 Unauthorised
VA_02	127 Interworking	407 Proxy authentication required
VA_03	127 Interworking	413 Request Entity too long
VA_04	127 Interworking	414 Request-uri too long
VA_05	127 Interworking	415 Unsupported Media type
VA_06	127 Interworking	416 Unsupported URI scheme
VA_07	127 Interworking	420 Bad Extension
VA_08	127 Interworking	421 Extension required
VA_09	28 Invalid Number format	484 Address Incomplete
VA_10	127 Interworking	503 Service Unavailable
VA_11	127 Interworking	505 Version not supported
VA_12	127 Interworking	513 Message too large
VA_13	127 Interworking	580 Precondition failure

TP308009	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.6
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/12	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a Failure message (4xx, 5xx, 6xx) where a Reason header field with ITU-T Rec Q.850 [5] Cause Value is not included defined as SIP_Failure_VA: <ul style="list-style-type: none"> • No action is taken on the ISUP. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM	<p style="text-align: center;">SUT</p> <pre> sequenceDiagram participant SUT participant ISUP SUT->>ISUP: SIP IAM ISUP->>SUT: SIP INVITE SUT->>ISUP: SIP Failure VA SUT->>ISUP: ACK </pre> <p>Further SIP procedures apply</p>

Table 23

VA	←4XX/5XX/6XX SIP message SIP_Failure_VA
VA_01	401 Unauthorised
VA_02	407 Proxy authentication required
VA_03	413 Request Entity too long
VA_04	414 Request-uri too long
VA_05	415 Unsupported Media type
VA_06	416 Unsupported URI scheme
VA_07	420 Bad Extension
VA_08	421 Extension required
VA_09	484 Address Incomplete
VA_10	503 Service Unavailable
VA_11	505 Version not supported
VA_12	513 Message too large
VA_13	580 Precondition failure

TP308010	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.6
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/11	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM sends out an INVITE message, on receipt of a Failure message 487 Request terminated where a Reason header field with ITU-T Rec Q.850 [5] Cause Value is not included: <ul style="list-style-type: none"> No action is taken on the ISUP if a CANCEL request was previously sent before answer to an INVITE 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM → SUT → SIP REL → ← INVITE RLC ← → 100 TRYING RLC ← → CANCEL RLC ← → 200 OK CANCEL	487 Request terminated ACK

TP308011	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.6
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/11	
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT after receiving the IAM sends out an INVITE message, on receipt of a Failure message 491 Request Pending where a Reason header field with ITU-T Rec Q.850 [5] Cause Value is not included:</p> <ul style="list-style-type: none"> No action is taken on the ISUP. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM	SUT → → ← → SIP INVITE 491 Request Pending ACK

TP308013	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.6
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/11	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM sends out an INVITE message, a SIP message defined as SIP MESSAGE_VA has been received, on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA where a Reason header field with ITU-T Rec Q.850 [5] Cause Value is not included: <ul style="list-style-type: none"> • sends a REL message with the Cause value set to CV_ISUP. 	
SIP Parameter values:		
ISUP Parameter values:	REL; cause value: CV_ISUP	
Comments:	ISUP IAM REL RLC	SUT → INVITE ← SIP MESSAGE_VA ← SIP_Failure_VA → ACK

Table 24

Values for test purpose TP308013	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress

Table 25

Values for test purposes TP308013 and TP308017		
VA	←REL (Cause Value) CV_ISUP	←4XX/5XX/6XX SIP message SIP_Failure_VA
VA_01	127 Interworking	400 Bad Request
VA_02	127 Interworking	402 Payment Required
VA_03	127 Interworking	403 Forbidden
VA_04	1 Unallocated number	404 Not Found
VA_05	127 Interworking	405 Method Not Allowed
VA_06	127 Interworking	406 Not Acceptable
VA_07	127 Interworking	408 Request Timeout
VA_08	22 Number changed (without diagnostic)	410 Gone
VA_09	127 Interworking	423 Interval Too Brief
VA_10	20 Subscriber absent	480 Temporarily Unavailable
VA_11	127 Interworking	481 Call/Transaction does not exist
VA_12	127 Interworking	482 Loop Detected
VA_13	127 Interworking	483 Too many hops
VA_14	127 Interworking	485 Ambiguous
VA_15	17 User busy	486 Busy Here
VA_16	127 Interworking	488 Not acceptable here
VA_17	No mapping.	491 Request Pending
VA_18	127 Interworking	493 Undecipherable
VA_19	127 Interworking	500 Server Internal error
VA_20	127 Interworking	501 Not implemented
VA_21	127 Interworking	502 Bad Gateway
VA_22	127 Interworking	504 Server timeout
VA_23	17 User busy	600 Busy Everywhere
VA_24	21 Call rejected	603 Decline
VA_25	1 Unallocated number	604 Does not exist anywhere
VA_26	127 Interworking	606 Not acceptable

TP308014	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.6
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/11	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM sends out an INVITE message a 180 ringing message has been received on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA where a Reason header field with ITU-T Rec Q.850 [5] Cause Value is not included: <ul style="list-style-type: none"> sends a REL message with the Cause value CV_ISUP. 	
SIP Parameter values:		
ISUP Parameter values:	REL; cause value: CV_ISUP	
Comments:	ISUP IAM REL RLC	SUT → → ← → SIP INVITE 180 Ringing SIP_Failure_VA ACK

Table 26

Values for test purposes TP308014		
VA	←REL (Cause Value) CV_ISUP	←4XX/5XX/6XX SIP message SIP_Failure_VA
VA_01	127 Interworking	408 Request timeout
VA_02	17 User busy	486 Busy Here
VA_03	17 User busy	600 Busy Everywhere
VA_04	21 Call rejected	603 Decline

TP308017	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.6
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM sends out an INVITE message, a SIP message defined as SIP_MESSAGE_VA has been received on receipt of a Failure message (4xx, 5xx, 6xx) defined as SIP_Failure_VA where a Reason header field with ITU-T Rec Q.850 [5] Cause Value is included: <ul style="list-style-type: none"> sends a REL message. The Cause Value in the header field set to CV_SIP is mapped to the ISUP Cause Value field in the ISUP REL message with the Cause value set to CV_ISUP. 	
SIP Parameter values:	CV_SIP (PIXIT)	
ISUP Parameter values:	CV_ISUP (PIXIT)	
Comments:	ISUP IAM REL RLC	SUT → ← → → SIP INVITE SIP MESSAGE_VA SIP_Failure_VA ACK

Table 27

Values for test purpose TP308017	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress

TP308018	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.6
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/17	
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a response message (3xx) defined as SIP_Response_VA, the SUT:</p> <ul style="list-style-type: none"> sends a REL message with the Cause value CV_ISUP. 	
SIP Parameter values:		
ISUP Parameter values:	REL; cause value: CV_ISUP	
Comments:	ISUP IAM → SUT → SIP REL ← → SIP_Response_VA RLC → → ACK	

Table 28

Values for test purposes TP308018		
VA	←REL (Cause Value) CV_ISUP	←3XX SIP message SIP_Response_VA
VA_01	127 Interworking	300 Multiple Choices
VA_02	127 Interworking	301 Moved Permanently
VA_03	127 Interworking	302 Move Temporarily
VA_04	127 Interworking	305 Use Proxy
VA_05	127 Interworking	380 Alternative Service

TP308019	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.6	
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/		
SIP selection criteria:	PICS 4/17		
ISUP selection criteria:			
Test purpose:	Ensure that the SUT after receiving the IAM sends out an INVITE message. On receipt of a response message (3xx) defined as SIP_Response_VA , the SUT: <ul style="list-style-type: none"> sends an INVITE using the value of the Contact header field in the received SIP_Response_VA in the Request URI. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP IAM → ↗ ACM ← ↙ ANM ← ↙ REL → ↗ RLC ← ↙	SUT → ↗ ← ↙ → ↗ → ↗ → ↗ → ↗ → ↗ → ↗ → ↗ → ↗ → ↗ → ↗ Conversation → ↗ → ↗ → ↗ → ↗ → ↗ → ↗ → ↗ → ↗	SIP INVITE SIP_Response_VA ACK INVITE 180 Ringing 200 OK INVITE ACK BYE 200 OK BYE

Table 29

Values for test purpose TP308019	
VA	SIP_Response_VA
VA_1	300 Multiple Choices
VA_2	301 Moved Permanently
VA_3	302 Move Temporarily
VA_4	305 Use Proxy
VA_5	380 Alternative Service

6.2.2.9 Autonomous release at O-IWU

TP308020	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.3
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	NOT PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM the BICC/ISUP procedures results in autonomous REL message from the SUT: <ul style="list-style-type: none"> then a BYE shall be sent on the SIP side. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM → SUT → SIP ACM ← ← INVITE ANM ← ← 180 Ringing → ← 200 OK INVITE Autonomous release at O-IWU REL ← → BYE RLC → ← 200 OK BYE	

TP308021	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.3
TSS reference:	ISUP-SIP /Basic call/ Sending of the Release message (REL)/	
SIP selection criteria:	PICS 4/10	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT after receiving the IAM the BICC/ISUP procedures results in autonomous REL message from the SUT: <ul style="list-style-type: none"> then a BYE shall be sent on the SIP side. The Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value of the REL message has to be on sent by the SIP side. 	
SIP Parameter values:	BYE cause value: CV_SIP (PIXIT)	
ISUP Parameter values:	REL; cause value: CV_ISUP (PIXIT)	
Comments:	ISUP IAM → SUT → SIP ACM ← ← INVITE ANM ← ← 180 Ringing ← ← 200 OK INVITE Autonomous release at O-IWU REL ← → BYE RLC ← → 200 OK BYE	

TP308022	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.6.1
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	PICS 3/2	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT a On receipt of a 484 Address Incomplete response for the current INVITE (i.e. there are no other pending INVITE transactions for this call), if the SUT is configured to propagate overlap signalling into the SIP network, the SUT: <ul style="list-style-type: none"> shall not send a REL message immediately and shall instead start timer T_{oiw3}. The REL message shall only be sent if T_{oiw3} expires. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM → SUT → SIP ← ← INVITE ← ← 484 Address incomplete → → ACK Start timer T_{oiw3} . . Timeout T_{oiw3} REL ← → RLC ← →	

TP308023	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.6.1
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	NOT PICS 3/4	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT a On receipt of a 484 Address Incomplete response for the current INVITE (i.e. there are no other pending INVITE transactions for this call), if the O-IWU is not configured to propagate overlap signalling into the SIP network then the timer shall not be started and the:	<ul style="list-style-type: none"> • REL shall be sent immediately to the BICC/ISUP network.
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM REL RLC	SUT → SIP INVITE ← → 484 Address incomplete ACK ← →

TP308024	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.3
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/	
SIP selection criteria:	PICS 4/5 AND PICS 4/15	
ISUP selection criteria:	PICS 4/2	
Test purpose:	Ensure that the SUT a on receipt of a COT "failed" and preconditions used, the SUT:	<ul style="list-style-type: none"> • sends a CANCEL or BYE to the SIP network.
SIP Parameter values:		
ISUP Parameter values:	IAM: Nature of connection indicators "continuity check required on this circuit"	
Comments:	ISUP IAM COT(failed) CASE A terminated CASE B	SUT → SIP INVITE → → → CANCEL → → 200 OK CANCEL → → 487 Request → → ACK → → BYE → → 200 OK BYE → → 487 Request terminated → → ACK

TP308025	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clause 7.7.3																								
TSS reference:	ISUP-SIP/Basic call/ Receipt of the Release message (REL)/																									
SIP selection criteria:	PICS 4/5 AND PICS 4/15																									
ISUP selection criteria:	PICS 4/2																									
Test purpose:	<p>Ensure that the SUT when the ISUP/BICC timer T8 is expired and preconditions used, the SUT:</p> <ul style="list-style-type: none"> sends a CANCEL or BYE to the SIP network. 																									
SIP Parameter values:																										
ISUP Parameter values:	IAM: Nature of connection indicators "continuity check required on this circuit"																									
Comments:	<p>ISUP IAM → SUT → SIP INVITE</p> <p style="text-align: center;">T8 expires</p> <table> <tr> <td>CASE A</td> <td>→</td> <td>CANCEL</td> </tr> <tr> <td></td> <td>←</td> <td>200 OK CANCEL</td> </tr> <tr> <td></td> <td>←</td> <td>487 Request</td> </tr> <tr> <td>terminated</td> <td>→</td> <td>ACK</td> </tr> <tr> <td>CASE B</td> <td>→</td> <td>BYE</td> </tr> <tr> <td></td> <td>←</td> <td>200 OK BYE</td> </tr> <tr> <td></td> <td>←</td> <td>487 Request terminated</td> </tr> <tr> <td></td> <td>→</td> <td>ACK</td> </tr> </table>		CASE A	→	CANCEL		←	200 OK CANCEL		←	487 Request	terminated	→	ACK	CASE B	→	BYE		←	200 OK BYE		←	487 Request terminated		→	ACK
CASE A	→	CANCEL																								
	←	200 OK CANCEL																								
	←	487 Request																								
terminated	→	ACK																								
CASE B	→	BYE																								
	←	200 OK BYE																								
	←	487 Request terminated																								
	→	ACK																								

6.2.2.10 Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented

6.2.2.10.1 Receipt of Reset Circuit message (RSC)

TP309001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 1), 7.7.4 and 7.7.5	
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	<p>Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of a RSC message:</p> <ul style="list-style-type: none"> • no action is required on the SIP side other than to terminate local procedures if any are in progress. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP IAM RSC RLC	SUT → → ←	SIP

TP309002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 7.7.4 and 7.7.5
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message before a SIP MESSAGE_VA response message has been received:</p> <ul style="list-style-type: none"> • The SUT shall hold the RSC message until a SIP response has been received. • The SUT shall send a CANCEL request. • Depending on local policy, a Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value # 31 may be added to the SIP message to be sent by the SIP side of the O-IWU. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM RSC RLC	SUT

Table 30

Values for test purpose TP309002	
VA	SIP MESSAGE_VA
VA_1	100 Trying
VA_2	180 Ringing
VA_3	181 Call Is Being Forwarded
VA_4	182 Queued
VA_5	183 Session Progress

TP309003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 7.7.4 and 7.7.5	
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message before a 200 OK response message has been received:</p> <ul style="list-style-type: none"> On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. Depending on local policy, a Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value # 31 may be added to the SIP message to be sent by the SIP side of the O-IWU. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP IAM RSC RLC	SUT 	SIP INVITE 200 OK INVITE ACK BYE 200 OK BYE

TP309005	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 7.7.4 and 7.7.5																								
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																									
SIP selection criteria:																										
ISUP selection criteria:																										
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a BYE message on receipt RSC message after a 200 OK response message has been received:</p> <ul style="list-style-type: none"> • The SUT shall send a BYE request. • Depending on local policy, a Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value # 31 may be added to the SIP message to be sent by the SIP side of the O-IWU. 																									
SIP Parameter values:																										
ISUP Parameter values:																										
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP</td> <td style="width: 25%; text-align: center;">→</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">SIP</td> </tr> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> <td>INVITE</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> <td>180 Ringing</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> <td>200 OK INVITE</td> </tr> <tr> <td>RSC</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> <td>BYE</td> </tr> <tr> <td>RLC</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> <td>200 OK BYE</td> </tr> </table>	ISUP	→	SUT	SIP	IAM	→	→	INVITE	ACM	←	←	180 Ringing	ANM	←	←	200 OK INVITE	RSC	→	→	BYE	RLC	←	←	200 OK BYE	
ISUP	→	SUT	SIP																							
IAM	→	→	INVITE																							
ACM	←	←	180 Ringing																							
ANM	←	←	200 OK INVITE																							
RSC	→	→	BYE																							
RLC	←	←	200 OK BYE																							

TP309006	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 7.7.4 and 7.7.5																
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																	
SIP selection criteria:																		
ISUP selection criteria:																		
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt RSC message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established:</p> <ul style="list-style-type: none"> • The SUT shall send a CANCEL or BYE request. • Depending on local policy, a Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value # 31 may be added to the SIP message to be sent by the SIP side of the O-IWU. 																	
SIP Parameter values:																		
ISUP Parameter values:																		
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP IAM</td> <td style="width: 25%; text-align: center;">→</td> <td style="width: 25%; text-align: center;">SUT →</td> <td style="width: 25%; text-align: center;">SIP INVITE</td> </tr> <tr> <td>RSC RLC</td> <td style="text-align: center;">→</td> <td style="text-align: center;">←</td> <td style="text-align: center;">SIP_MESSAGE_VA</td> </tr> <tr> <td>Case A</td> <td></td> <td></td> <td></td> </tr> <tr> <td>Case B</td> <td></td> <td></td> <td></td> </tr> </table>		ISUP IAM	→	SUT →	SIP INVITE	RSC RLC	→	←	SIP_MESSAGE_VA	Case A				Case B			
ISUP IAM	→	SUT →	SIP INVITE															
RSC RLC	→	←	SIP_MESSAGE_VA															
Case A																		
Case B																		

Table 31

Values for test purpose; TP309106	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress

6.2.2.10.2 Receipt of Circuit group reset message (GRS)

TP309007	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 1), 7.7 and 7.7.5
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of GRS message:</p> <ul style="list-style-type: none"> No action is required on the SIP side other than to terminate local procedures if any are in progress. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM GRS GRA	SUT → → ← SIP

TP309008	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 7.7.4 and 7.7.5
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message before SIP MESSAGE_VA response message has been received:</p> <ul style="list-style-type: none"> The SUT shall hold the GRS message until a SIP response has been received. The SUT shall send a CANCEL request. Depending on local policy, a Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value # 31 may be added to the SIP message to be sent by the SIP side of the O-IWU. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM GRS GRA	SUT → → ← SIP INVITE ← SIP MESSAGE_VA → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK

Table 32

Values for test purpose TP309008	
VA	SIP MESSAGE_VA
VA_1	100 Trying
VA_2	180 Ringing
VA_3	181 Call Is Being Forwarded
VA_4	182 Queued
VA_5	183 Session Progress

TP309009	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 3), 7.7.4 and 7.7.5	
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message before a 200 OK response message has been received</p> <ul style="list-style-type: none"> The SUT shall hold the GRS message until a response has been received. A CANCEL is sent. On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	ISUP IAM → GRS → GRA ←	SUT → ← → ← → ← → ← → ←	SIP INVITE 100 TRYIING CANCEL 200 OK INVITE ACK 200 OK CANCEL BYE 200 OK BYE

TP309011	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 7.7.4 and 7.7.5																								
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																									
SIP selection criteria:																										
ISUP selection criteria:																										
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a BYE message on receipt GRS message after a 200 OK response message has been received:</p> <ul style="list-style-type: none"> • The SUT shall send a BYE request. • Depending on local policy, a Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value # 31 may be added to the SIP message to be sent by the SIP side of the O-IWU. 																									
SIP Parameter values:																										
ISUP Parameter values:																										
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP</td> <td style="width: 25%; text-align: center;">→</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">SIP</td> </tr> <tr> <td>IAM</td> <td>→</td> <td></td> <td>INVITE</td> </tr> <tr> <td>ACM</td> <td>←</td> <td>←</td> <td>180 Ringing</td> </tr> <tr> <td>ANM</td> <td>←</td> <td>←</td> <td>200 OK INVITE</td> </tr> <tr> <td>GRS</td> <td>→</td> <td>→</td> <td>BYE</td> </tr> <tr> <td>GRA</td> <td>←</td> <td>←</td> <td>200 OK BYE</td> </tr> </table>	ISUP	→	SUT	SIP	IAM	→		INVITE	ACM	←	←	180 Ringing	ANM	←	←	200 OK INVITE	GRS	→	→	BYE	GRA	←	←	200 OK BYE	
ISUP	→	SUT	SIP																							
IAM	→		INVITE																							
ACM	←	←	180 Ringing																							
ANM	←	←	200 OK INVITE																							
GRS	→	→	BYE																							
GRA	←	←	200 OK BYE																							

TP309012	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 7.7.4 and 7.7.5																				
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																					
SIP selection criteria:																						
ISUP selection criteria:																						
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt GRS message after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established:</p> <ul style="list-style-type: none"> • The SUT shall send a CANCEL request. • Depending on local policy, a Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value # 31 may be added to the SIP message to be sent by the SIP side of the O-IWU. 																					
SIP Parameter values:																						
ISUP Parameter values:																						
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 30%;">ISUP IAM</td> <td style="width: 10%; text-align: center;">→</td> <td style="width: 30%;">SUT</td> <td style="width: 10%; text-align: center;">→</td> <td style="width: 20%;">SIP INVITE</td> </tr> <tr> <td>GRS GRA</td> <td style="text-align: center;">→</td> <td></td> <td style="text-align: center;">←</td> <td>SIP_MESSAGE_VA</td> </tr> <tr> <td>CASE A</td> <td></td> <td></td> <td></td> <td> → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK </td> </tr> <tr> <td>CASE B</td> <td></td> <td></td> <td></td> <td> → BYE ← 200 OK BYE ← 487 Request terminated → ACK </td> </tr> </table>		ISUP IAM	→	SUT	→	SIP INVITE	GRS GRA	→		←	SIP_MESSAGE_VA	CASE A				→ CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK	CASE B				→ BYE ← 200 OK BYE ← 487 Request terminated → ACK
ISUP IAM	→	SUT	→	SIP INVITE																		
GRS GRA	→		←	SIP_MESSAGE_VA																		
CASE A				→ CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK																		
CASE B				→ BYE ← 200 OK BYE ← 487 Request terminated → ACK																		

Table 33

Values for test purpose TP309009 and TP309012	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress

TP309013	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 7.7.4 and 7.7.5																																																																
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																																																																	
SIP selection criteria:																																																																		
ISUP selection criteria:																																																																		
Test purpose:	<p>Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a GRS message where the Range Parameter value is bigger than "1":</p> <ul style="list-style-type: none"> • the SUT shall send a BYE requests for each call association. 																																																																	
SIP Parameter values:																																																																		
ISUP Parameter values:																																																																		
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 20%;">ISUP</td> <td style="width: 20%; text-align: center;">→</td> <td style="width: 20%; text-align: center;">SUT</td> <td style="width: 20%; text-align: center;">→</td> <td style="width: 20%;">SIP</td> </tr> <tr> <td>IAM</td> <td>→</td> <td></td> <td>→</td> <td>INVITE 1</td> </tr> <tr> <td>ACM</td> <td>←</td> <td></td> <td>←</td> <td>180 Ringing</td> </tr> <tr> <td>ANM</td> <td>←</td> <td></td> <td>←</td> <td>200 OK INVITE</td> </tr> <tr><td colspan="5"> </td></tr> <tr> <td>IAM</td> <td>→</td> <td></td> <td>→</td> <td>INVITE 2</td> </tr> <tr> <td>ACM</td> <td>←</td> <td></td> <td>←</td> <td>180 Ringing</td> </tr> <tr> <td>ANM</td> <td>←</td> <td></td> <td>←</td> <td>200 OK INVITE</td> </tr> <tr><td colspan="5"> </td></tr> <tr> <td>GRS</td> <td>→</td> <td></td> <td>→</td> <td>BYE 1</td> </tr> <tr> <td>GRA</td> <td>←</td> <td></td> <td>←</td> <td>200 OK BYE</td> </tr> <tr> <td></td> <td></td> <td></td> <td>→</td> <td>BYE 2</td> </tr> <tr> <td></td> <td></td> <td></td> <td>←</td> <td>200 OK BYE</td> </tr> </table>	ISUP	→	SUT	→	SIP	IAM	→		→	INVITE 1	ACM	←		←	180 Ringing	ANM	←		←	200 OK INVITE	 					IAM	→		→	INVITE 2	ACM	←		←	180 Ringing	ANM	←		←	200 OK INVITE	 					GRS	→		→	BYE 1	GRA	←		←	200 OK BYE				→	BYE 2				←	200 OK BYE
ISUP	→	SUT	→	SIP																																																														
IAM	→		→	INVITE 1																																																														
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GRA	←		←	200 OK BYE																																																														
			→	BYE 2																																																														
			←	200 OK BYE																																																														

6.2.2.10.3 Receipt of Circuit group blocking message (CGB)

TP309014	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 1) and 7.7.4																			
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																				
SIP selection criteria:																					
ISUP selection criteria:																					
Test purpose:	<p>Ensure that the SUT after receiving the IAM but before an INVITE has been sent on receipt of CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented":</p> <ul style="list-style-type: none"> • No action is required on the SIP side other than to terminate local procedures if any are in progress. 																				
SIP Parameter values:																					
ISUP Parameter values:																					
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 20%;">ISUP</td> <td style="width: 20%; text-align: center;">→</td> <td style="width: 20%; text-align: center;">SUT</td> <td style="width: 20%; text-align: center;">→</td> <td style="width: 20%;">SIP</td> </tr> <tr> <td>IAM</td> <td>→</td> <td></td> <td>→</td> <td></td> </tr> <tr> <td>CGB</td> <td>→</td> <td></td> <td>→</td> <td></td> </tr> <tr> <td>CGBA</td> <td>←</td> <td></td> <td>←</td> <td></td> </tr> </table>	ISUP	→	SUT	→	SIP	IAM	→		→		CGB	→		→		CGBA	←		←	
ISUP	→	SUT	→	SIP																	
IAM	→		→																		
CGB	→		→																		
CGBA	←		←																		

TP309015	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1 and 7.7.4
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" before a SIP MESSAGE_VA response message has been received:</p> <ul style="list-style-type: none"> The SUT shall hold the CGB message until a SIP 200 OK response has been received. The SUT shall send a CANCEL request. Depending on local policy, a Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value # 31 may be added to the SIP message to be sent by the SIP side of the O-IWU. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM → CGB → CGBA ←	SUT → INVITE ← SIP MESSAGE_VA → CANCEL ← 200 OK CANCEL ← 487 Request terminated → ACK

Table 34

Values for test purpose TP309014	
VA	SIP MESSAGE_VA
VA_1	100 Trying
VA_2	180 Ringing
VA_3	181 Call Is Being Forwarded
VA_4	182 Queued
VA_5	183 Session Progress

TP309016	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 3 and 7.7.4
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" before a 200 OK response message has been received:</p> <ul style="list-style-type: none"> On subsequently receiving 200 OK INVITE messages, the SUT shall send an ACK for the 200 OK INVITE and subsequently send a BYE request after the ACK has been sent. Depending on local policy, a Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value # 31 may be added to the SIP message to be sent by the SIP side of the O-IWU. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP IAM → CGB → CGBA ←	SUT → → 100 TRYING → CANCEL ← 200 OK INVITE → ACK ← 200 OK CANCEL → BYE ← 200 OK BYE

TP309017	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1 and 7.7.4																							
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																								
SIP selection criteria:																									
ISUP selection criteria:																									
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message with the complete called party number, sending a BYE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" after a 200 OK response message has been received:</p> <ul style="list-style-type: none"> • The SUT shall send a BYE request. • Depending on local policy, a Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value # 31 may be added to the SIP message to be sent by the SIP side of the O-IWU. 																								
SIP Parameter values:																									
ISUP Parameter values:																									
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 25%;">ISUP</th> <th style="text-align: center; width: 50%;">SUT</th> <th style="text-align: right; width: 25%;">SIP</th> </tr> </thead> <tbody> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> <td>INVITE</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> <td>180 Ringing</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> <td>200 OK INVITE</td> </tr> <tr> <td>CGB</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> <td>BYE</td> </tr> <tr> <td>CGBA</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> <td>200 OK BYE</td> </tr> </tbody> </table>	ISUP	SUT	SIP	IAM	→	→	INVITE	ACM	←	←	180 Ringing	ANM	←	←	200 OK INVITE	CGB	→	→	BYE	CGBA	←	←	200 OK BYE	
ISUP	SUT	SIP																							
IAM	→	→	INVITE																						
ACM	←	←	180 Ringing																						
ANM	←	←	200 OK INVITE																						
CGB	→	→	BYE																						
CGBA	←	←	200 OK BYE																						

TP309018	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1 and 7.7.4
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT after receiving the IAM with the complete called party number, sending a INVITE message on receipt CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" after an early dialogue with the SIP message defined with the SIP_MESSAGE_VA has been established:</p> <ul style="list-style-type: none"> • The SUT shall send a CANCEL request. • Depending on local policy, a Reason header field containing the (ITU-T Rec Q.850 [5]) Cause Value # 31 may be added to the SIP message to be sent by the SIP side of the O-IWU. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<pre> sequenceDiagram participant SUT participant SIP SUT->>SIP: IAM SIP->>SUT: INVITE SIP->>SUT: SIP_MESSAGE_VA SUT->>SIP: CGB SIP->>SUT: CANCEL SIP->>SUT: 200 OK CANCEL SIP->>SUT: 487 Request terminated SIP->>SUT: ACK SUT->>SIP: CASE A SIP->>SUT: BYE SIP->>SUT: 200 OK BYE SIP->>SUT: 487 Request terminated SIP->>SUT: ACK </pre>	

Table 35

Values for test purpose TP309014; TP309018	
VA	SIP MESSAGE_VA
VA_1	180 Ringing
VA_2	181 Call Is Being Forwarded
VA_3	182 Queued
VA_4	183 Session Progress

TP309019	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], clauses 7.7.1, 7.7.4 and 7.7.5																																																				
TSS reference:	ISUP-SIP/Basic call/ Receipt of Reset circuit message (RSC), Circuit group reset message (GRS) or Circuit group blocking message (CGB) with the indication hardware failure oriented																																																					
SIP selection criteria:																																																						
ISUP selection criteria:																																																						
Test purpose:	<p>Ensure that the SUT after receiving more than one IAM's sending an INVITE message for each call association on receipt of a CGB message Circuit Group Supervision Message Type Indicator coded as "hardware failure oriented" where the Range and Status Parameter value is bigger than "1".</p> <ul style="list-style-type: none"> • the SUT shall send a BYE requests for each call association. 																																																					
SIP Parameter values:																																																						
ISUP Parameter values:																																																						
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP</td> <td style="width: 25%; text-align: center;">→</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">SIP</td> </tr> <tr> <td>IAM</td> <td style="text-align: center;">←</td> <td style="text-align: center;">→</td> <td>INVITE 1</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> <td>180 Ringing</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> <td>200 OK INVITE</td> </tr> <tr> <td colspan="4"> </td> </tr> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> <td>INVITE 2</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> <td>180 Ringing</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> <td>200 OK INVITE</td> </tr> <tr> <td colspan="4"> </td> </tr> <tr> <td>CGB</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> <td>BYE 1</td> </tr> <tr> <td>CGBA</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> <td>200 OK BYE</td> </tr> <tr> <td></td> <td></td> <td style="text-align: center;">→</td> <td>BYE 2</td> </tr> <tr> <td></td> <td></td> <td style="text-align: center;">←</td> <td>200 OK BYE</td> </tr> </table>	ISUP	→	SUT	SIP	IAM	←	→	INVITE 1	ACM	←	←	180 Ringing	ANM	←	←	200 OK INVITE	 				IAM	→	→	INVITE 2	ACM	←	←	180 Ringing	ANM	←	←	200 OK INVITE	 				CGB	→	→	BYE 1	CGBA	←	←	200 OK BYE			→	BYE 2			←	200 OK BYE	
ISUP	→	SUT	SIP																																																			
IAM	←	→	INVITE 1																																																			
ACM	←	←	180 Ringing																																																			
ANM	←	←	200 OK INVITE																																																			
IAM	→	→	INVITE 2																																																			
ACM	←	←	180 Ringing																																																			
ANM	←	←	200 OK INVITE																																																			
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CGBA	←	←	200 OK BYE																																																			
		→	BYE 2																																																			
		←	200 OK BYE																																																			

6.3 Test purposes for the Supplementary Services

6.3.1 Interworking from SIP to ISUP (Outgoing Call)

6.3.1.1 Calling Line Identification (CLI)

TP501001	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]	
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection criteria:			
ISUP selection criteria:	PICS 6/1 AND PICS 6/9		
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • a Privacy header field has not been received. <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 20px;">Address signals = default number</p> <p style="margin-left: 20px;">Screening indicator = network provided</p> <p style="margin-left: 20px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 20px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 20px;">Address Presentation Restricted Indicator = Presentation allowed</p> <p style="margin-left: 20px;">NoAS: NoA_VALUE</p>		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE	SUT →	ISUP IAM

Table 36

Values for test purposes TP501001			
		SIP Parameter values:	ISUP Parameter value Address Format:
VA_01	NoAS_VALUE: " <i>national (significant) number</i> "	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: " <i>international number</i> " ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501002	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/1	
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • a Privacy header field was received and the priv-value component is set to "none". <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 40px;">Address signals = default number</p> <p style="margin-left: 40px;">Screening indicator = network provided</p> <p style="margin-left: 40px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 40px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 40px;">Address Presentation Restricted Indicator = Presentation allowed</p> <p style="margin-left: 40px;">NoAS: NoA_VALUE</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<ul style="list-style-type: none"> • If CC is equal to the country code of the country where I-IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number"; • else set to "international number". 	
SIP INVITE	→	SUT
		→
		ISUP IAM

Table 37

Values for test purposes TP501002			
		SIP Parameter values:	ISUP Parameter value Address Format:
VA_01	NoAS_VALUE: "national (significant) number"	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501003	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]	
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection criteria:			
ISUP selection criteria:	PICS 6/1		
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • a Privacy header field was received and the priv-value component is set to "header". <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 40px;">Address signals = default number</p> <p style="margin-left: 40px;">Screening indicator = network provided</p> <p style="margin-left: 40px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 40px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 40px;">Address Presentation Restricted Indicator = Presentation restricted</p>		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE	SUT →	ISUP IAM

TP501004	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]	
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection criteria:			
ISUP selection criteria:	PICS 6/1		
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • a Privacy header field was received and the priv-value component is set to "user". <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 40px;">Address signals = default number</p> <p style="margin-left: 40px;">Screening indicator = network provided</p> <p style="margin-left: 40px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 40px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 40px;">Address Presentation Restricted Indicator = Presentation restricted</p>		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE	SUT →	ISUP IAM

TP501005	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/1	
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • a Privacy header field was received and the priv-value component is set to "id". <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 40px;">Address signals = default number</p> <p style="margin-left: 40px;">Screening indicator = network provided</p> <p style="margin-left: 40px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 40px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 40px;">Address Presentation Restricted Indicator = Presentation restricted</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP INVITE	SUT → ISUP IAM

TP501006	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]		
TSS reference:	SIP-ISUP/SS/CLI/			
SIP selection criteria:				
ISUP selection criteria:	PICS 6/1 AND PICS 6/3 AND PICS 6/9			
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field has not been received. <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 20px;">Address signals = default number</p> <p style="margin-left: 20px;">Screening indicator = network provided</p> <p style="margin-left: 20px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 20px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 20px;">Address Presentation Restricted Indicator = Presentation allowed</p> <p style="margin-left: 20px;">NoAS: NoA_VALUE</p> <p>with the Generic number parameter coded:</p> <p style="margin-left: 20px;">Address signals = number provided by the user</p> <p style="margin-left: 20px;">Screening indicator = user provided, not verified</p> <p style="margin-left: 20px;">Number Incomplete Indicator = complete</p> <p style="margin-left: 20px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 20px;">Address Presentation Restricted Indicator = Presentation allowed</p> <p style="margin-left: 20px;">NoAS: NoA_VALUE</p>			
SIP Parameter values:				
ISUP Parameter values:				
Comments:	<ul style="list-style-type: none"> • If CC is equal to the country code of the country where I-IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number"; • else set to "international number". 			
SIP INVITE	→	SUT	→	ISUP IAM

Table 38

Values for test purposes TP501006			
		SIP Parameter values:	ISUP Parameter value Address Format:
VA_0 1	NoAS_VALUE: "national (significant) number"	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_0 2	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501007	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/1 AND PICS 6/3	
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; a Privacy header field was received and the priv-value component is set to "none". <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="padding-left: 20px;">Address signals = default number</p> <p style="padding-left: 20px;">Screening indicator = network provided</p> <p style="padding-left: 20px;">Number Incomplete Indicator = PIXIT</p> <p style="padding-left: 20px;">Numbering plan indicator = ISDN numbering plan</p> <p style="padding-left: 20px;">Address Presentation Restricted Indicator = Presentation allowed</p> <p style="padding-left: 20px;">NoAS: NoA_VALUE</p> <p>with the Generic number parameter coded:</p> <p style="padding-left: 20px;">Address signals = number provided by the user</p> <p style="padding-left: 20px;">Screening indicator = user provided, not verified</p> <p style="padding-left: 20px;">Number Incomplete Indicator = complete</p> <p style="padding-left: 20px;">Numbering plan indicator = ISDN numbering plan</p> <p style="padding-left: 20px;">Address Presentation Restricted Indicator = Presentation allowed</p> <p style="padding-left: 20px;">NoAS: NoA_VALUE</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP INVITE	SUT → ISUP IAM

Table 39

Values for test purposes TP501007			
	SIP Parameter values:	ISUP Parameter value Address Format:	
VA_01	NoAS_VALUE: "national (significant) number"	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501008	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]	
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection criteria:			
ISUP selection criteria:	PICS 6/1 AND PICS 6/3		
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field was received and the priv-value component is set to "header". <p>sends an IAM message with the Calling party number parameter coded:</p> <p>Address signals = default number</p> <p>Screening indicator = network provided</p> <p>Number Incomplete Indicator = PIXIT</p> <p>Numbering plan indicator = ISDN numbering plan</p> <p>Address Presentation Restricted Indicator = Presentation restricted</p> <p>NoAS: NoA_VALUE</p> <p>with the Generic number parameter coded:</p> <p>Address signals = number provided by the user</p> <p>Screening indicator = user provided, not verified</p> <p>Number Incomplete Indicator = complete</p> <p>Numbering plan indicator = ISDN numbering plan</p> <p>Address Presentation Restricted Indicator = Presentation restricted</p> <p>NoAS: NoA_VALUE</p>		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	<ul style="list-style-type: none"> • If CC is equal to the country code of the country where I-IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number"; • else set to "international number". 		
	SIP INVITE	SUT →	ISUP IAM

Table 40

Values for test purposes TP501008			
		SIP Parameter values:	ISUP Parameter value Address Format:
VA_01	NoAS_VALUE: "national (significant) number"	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501009	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/1 AND PICS 6/3	
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field was received and the priv-value component is set to "user". <p>sends an IAM message with the Calling party number parameter coded:</p> <p>Address signals = default number Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE</p> <p>with the Generic number parameter coded:</p> <p>Address signals = number provided by the user Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<ul style="list-style-type: none"> • If CC is equal to the country code of the country where I-IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number"; • else set to "international number". 	
	SIP INVITE	SUT
		ISUP IAM

Table 41

Values for test purposes TP501009			
	SIP Parameter values:	ISUP Parameter value Address Format:	
VA_01	NoAS_VALUE: "national (significant) number"	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501010	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/1 AND PICS 6/3	
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field was received and the priv-value component is set to "id". <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 20px;">Address signals = default number</p> <p style="margin-left: 20px;">Screening indicator = network provided</p> <p style="margin-left: 20px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 20px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 20px;">Address Presentation Restricted Indicator = Presentation restricted</p> <p style="margin-left: 20px;">NoAS: NoA_VALUE</p> <p>with the Generic number parameter coded:</p> <p style="margin-left: 20px;">Address signals = number provided by the user</p> <p style="margin-left: 20px;">Screening indicator = user provided, not verified</p> <p style="margin-left: 20px;">Number Incomplete Indicator = complete</p> <p style="margin-left: 20px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 20px;">Address Presentation Restricted Indicator = Presentation restricted</p> <p style="margin-left: 20px;">NoAS: NoA_VALUE</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<ul style="list-style-type: none"> • If CC is equal to the country code of the country where I-IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number"; • else set to "international number". 	
	SIP INVITE	SUT
		ISUP IAM

Table 42

Values for test purposes TP501010			
		SIP Parameter values:	ISUP Parameter value Address Format:
VA_01	NoAS_VALUE: "national (significant) number"	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501011	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; a Privacy header field has not been received. <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 40px;">Address signals = number derived from SIP P-Asserted-Identity</p> <p style="margin-left: 40px;">Screening indicator = network provided</p> <p style="margin-left: 40px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 40px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 40px;">Address Presentation Restricted Indicator = Presentation allowed</p> <p style="margin-left: 40px;">NoAS: NoA_VALUE</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<ul style="list-style-type: none"> If CC is equal to the country code of the country where I-IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number"; else set to "international number". 	SIP INVITE → SUT → ISUP IAM

Table 43

Values for test purposes TP501011			
		SIP Parameter values:	ISUP Parameter value Address Format:
VA_01	NoAS_VALUE: "national (significant) number"	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501012	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; a Privacy header field was received and the priv-value component is set to "none". <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 40px;">Address signals = number derived from SIP P-Asserted-Identity</p> <p style="margin-left: 40px;">Screening indicator = network provided</p> <p style="margin-left: 40px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 40px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 40px;">Address Presentation Restricted Indicator = Presentation allowed</p> <p style="margin-left: 40px;">NoAS: NoA_VALUE</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<ul style="list-style-type: none"> If CC is equal to the country code of the country where I-IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number"; else set to "international number". 	
	SIP INVITE	SUT
		ISUP IAM

Table 44

Values for test purposes TP501012			
	SIP Parameter values:	ISUP Parameter value Address Format	
VA_01	NoAS_VALUE: " <i>national (significant) number</i> " ("+" CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: " <i>international number</i> " ("+" CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501013	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • a Privacy header field was received and the priv-value component is set to "header". <p>sends an IAM message with the Calling party number parameter coded:</p> <p>Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP INVITE	SUT →
		ISUP IAM

TP501014	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • a Privacy header field was received and the priv-value component is set to "user". <p>sends an IAM message with the Calling party number parameter coded:</p> <p>Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP INVITE	SUT → ISUP IAM

TP501015	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • a Privacy header field was received and the priv-value component is set to "id". <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 40px;">Address signals = number derived from SIP P-Asserted-Identity</p> <p style="margin-left: 40px;">Screening indicator = network provided</p> <p style="margin-left: 40px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 40px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 40px;">Address Presentation Restricted Indicator = Presentation restricted</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP INVITE	SUT → ISUP IAM

Table 45: Void.

TP501016	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]		
TSS reference:	SIP-ISUP/SS/CLI/			
SIP selection criteria:				
ISUP selection criteria:	PICS 6/3			
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field has not been received. <p>sends an IAM message with the Calling party number parameter coded:</p> <p>Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE</p> <p>with the Generic number parameter coded:</p> <p>Address signals = number provided by the user Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE</p>			
SIP Parameter values:				
ISUP Parameter values:				
Comments:	<ul style="list-style-type: none"> • If CC is equal to the country code of the country where I-IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number"; • else set to "international number". 			
SIP INVITE	→	SUT	→	ISUP IAM

Table 46

Values for test purposes TP501016			
		SIP Parameter values:	ISUP Parameter value Address Format:
VA_01	NoAS_VALUE: " <i>national (significant) number</i> "	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: " <i>international number</i> " ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501017	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]		
TSS reference:	SIP-ISUP/SS/CLI/			
SIP selection criteria:				
ISUP selection criteria:	PICS 6/3			
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field was received and the priv-value component is set to "none". <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 20px;">Address signals = number derived from SIP P-Asserted-Identity</p> <p style="margin-left: 20px;">Screening indicator = network provided</p> <p style="margin-left: 20px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 20px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 20px;">Address Presentation Restricted Indicator = Presentation allowed</p> <p style="margin-left: 20px;">NoAS: NoA_VALUE</p> <p>with the Generic number parameter coded:</p> <p style="margin-left: 20px;">Address signals = number provided by the user</p> <p style="margin-left: 20px;">Screening indicator = user provided, not verified</p> <p style="margin-left: 20px;">Number Incomplete Indicator = complete</p> <p style="margin-left: 20px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 20px;">Address Presentation Restricted Indicator = Presentation allowed</p> <p style="margin-left: 20px;">NoAS: NoA_VALUE</p>			
SIP Parameter values:				
ISUP Parameter values:				
Comments:	<ul style="list-style-type: none"> • If CC is equal to the country code of the country where I-IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number"; • else set to "international number". 			
SIP INVITE	→	SUT	→	ISUP IAM

Table 47

Values for test purposes TP501017			
		SIP Parameter values:	ISUP Parameter value Address Format:
VA_01	NoAS_VALUE: "national (significant) number"	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501018	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/3	
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field was received and the priv-value component is set to "header". <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 20px;">Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE</p> <p>with the Generic number parameter coded:</p> <p style="margin-left: 20px;">Address signals = number provided by the user Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<ul style="list-style-type: none"> • If CC is equal to the country code of the country where I-IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number"; • else set to "international number". 	SIP INVITE → SUT → ISUP IAM

Table 48

Values for test purposes TP501018			
		SIP Parameter values:	ISUP Parameter value Address Format:
VA_01	NoAS_VALUE: "national (significant) number"	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501019	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]		
TSS reference:	SIP-ISUP/SS/CLI/			
SIP selection criteria:				
ISUP selection criteria:	PICS 6/3			
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field was received and the priv-value component is set to "user". <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 20px;">Address signals = number derived from SIP P-Asserted-Identity</p> <p style="margin-left: 20px;">Screening indicator = network provided</p> <p style="margin-left: 20px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 20px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 20px;">Address Presentation Restricted Indicator = Presentation restricted</p> <p style="margin-left: 20px;">NoAS: NoA_VALUE</p> <p>with the Generic number parameter coded:</p> <p style="margin-left: 20px;">Address signals = number provided by the user</p> <p style="margin-left: 20px;">Screening indicator = user provided, not verified</p> <p style="margin-left: 20px;">Number Incomplete Indicator = complete</p> <p style="margin-left: 20px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 20px;">Address Presentation Restricted Indicator = Presentation restricted</p> <p style="margin-left: 20px;">NoAS: NoA_VALUE</p>			
SIP Parameter values:				
ISUP Parameter values:				
Comments:	<ul style="list-style-type: none"> • If CC is equal to the country code of the country where I-IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number"; • else set to "international number". 			
SIP INVITE	→	SUT	→	ISUP IAM

Table 49

Values for test purposes TP501019			
	SIP Parameter values:	ISUP Parameter value Address Format:	
VA_01	NoAS_VALUE: "national (significant) number"	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501020	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/3	
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field was received and the priv-value component is set to "id". <p>sends an IAM message with the Calling party number parameter coded:</p> <p>Address signals = number derived from SIP P-Asserted-Identity Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE</p> <p>with the Generic number parameter coded:</p> <p>Address signals = number provided by the user Screening indicator = user provided, not verified Number Incomplete Indicator = complete Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation restricted NoAS: NoA_VALUE</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<ul style="list-style-type: none"> • If CC is equal to the country code of the country where I-IWU is located AND the next BICC/ISUP node is located in the same country then set to "national (significant) number"; • else set to "international number". 	
	SIP INVITE	SUT
	→	→
		ISUP IAM

Table 50

Values for test purposes TP501020			
		SIP Parameter values:	ISUP Parameter value Address Format:
VA_01	NoAS_VALUE: "national (significant) number"	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	NDC+SN
VA_02	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	From, userinfo component of URI assumed to be in form "+" CC+NDC+SN	CC+NDC+SN

TP501021	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 6/1 AND PICS 6/11	
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • a Privacy header field has not been received. <p>sends an IAM message with the Calling party number parameter coded:</p> <p>Address signals = absent Screening indicator = network provided Nature of address indicator = 0000000 Number Incomplete Indicator = 0 Numbering plan indicator = 000 Address Presentation Restricted Indicator = Address not available</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP INVITE	SUT → ISUP IAM

TP501022	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]
TSS reference:	SIP-ISUP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:	PICS 1/9	
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a SIP URI with an identity 1 in the format "+" CC+ NDC+ SN has been received without user = phone; • the SIP P-Asserted-Identity containing a Tel URI with an identity 2 in the format "+" CC+ NDC+ SN has been received; • a Privacy header field has not been received. <p>sends an IAM message with the Calling party number parameter coded:</p> <p>Address signals = identity 2 Screening indicator = network provided Number Incomplete Indicator = PIXIT Numbering plan indicator = ISDN numbering plan Address Presentation Restricted Indicator = Presentation allowed NoAS: NoA_VALUE</p>	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP INVITE	SUT → ISUP IAM

TP501023	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]	
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection criteria:			
ISUP selection criteria:	PICS 1/9 AND PICS 6/1 AND PICS 6/12		
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • a Privacy header field has not been received. <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 40px;">Address signals = default number</p> <p style="margin-left: 40px;">Screening indicator = network provided</p> <p style="margin-left: 40px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 40px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 40px;">Address Presentation Restricted Indicator = Presentation restricted by the network</p> <p style="margin-left: 40px;">NoAS: NoA_VALUE</p>		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE	SUT →	ISUP IAM

TP501024	SIP reference: RFC 3261 [6]	ISUP reference: 6.1.3.6 [2]	
TSS reference:	SIP-ISUP/SS/CLI/		
SIP selection criteria:			
ISUP selection criteria:	PICS 1/9 AND PICS 6/1 AND PICS 6/3 AND PICS 6/12		
Test purpose:	<p>Ensure that the SUT in the Idle state, on receipt of a INVITE message where:</p> <ul style="list-style-type: none"> • the SIP P-Asserted-Identity containing a URI with an identity in the format "+" CC+ NDC+ SN has not been received; • the SIP From header field containing a URI with an identity in the format "+" CC+ NDC+ SN has been received; • a Privacy header field has not been received. <p>sends an IAM message with the Calling party number parameter coded:</p> <p style="margin-left: 40px;">Address signals = default number</p> <p style="margin-left: 40px;">Screening indicator = network provided</p> <p style="margin-left: 40px;">Number Incomplete Indicator = PIXIT</p> <p style="margin-left: 40px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 40px;">Address Presentation Restricted Indicator = Presentation restricted by the network</p> <p style="margin-left: 40px;">NoAS: NoA_VALUE</p> <p>with the Generic number parameter coded:</p> <p style="margin-left: 40px;">Address signals = number provided by the user</p> <p style="margin-left: 40px;">Screening indicator = user provided, not verified</p> <p style="margin-left: 40px;">Number Incomplete Indicator = complete</p> <p style="margin-left: 40px;">Numbering plan indicator = ISDN numbering plan</p> <p style="margin-left: 40px;">Address Presentation Restricted Indicator = Presentation allowed</p> <p style="margin-left: 40px;">NoAS: NoA_VALUE</p>		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE	SUT →	ISUP IAM

Table 51

Values for test purposes TP501122, TP501023, TP501024			
	Nature of address indicator	SIP Parameter values:	ISUP Parameter value Address Format
VA_0 1	NoAS_VALUE: "national (significant) number"	CC contained in the P-Asserted-Identity is equal to the country where the I-IWU is located and the next BICC/ISUP node is in the same country	NDC+SN
VA_0 2	NoAS_VALUE: "international number" ("+"CC+NDC+SN)	CC contained in the P-Asserted-Identity is not equal to the country where the I-IWU is located or the next BICC/ISUP node is not in the same country	CC+NDC+SN

6.3.1.2 Call Hold (HOLD)

TP502001	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10																																				
TSS reference:	SIP-ISUP/SS/HOLD/																																					
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams.																																					
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service.																																					
Test purpose:	Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold. <ul style="list-style-type: none"> • The calling party should be able to put the other party on hold • The calling party should be able to retrieve the other party • The called party should be able to put the other party on hold • The called party should be able to retrieve the other party 																																					
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>																																					
ISUP Parameter values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)																																					
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP</th> <th style="text-align: center; width: 10%;">MGCF</th> <th style="text-align: right; width: 60%;">ISUP</th> </tr> </thead> <tbody> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">IAM</td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: right;">ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">ANM</td> </tr> <tr> <td>INVITE(sendonly)</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ CPG(hold)</td> </tr> <tr> <td>200 OK INVITE(recvonly)</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>INVITE(sendrecv)</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ CPG(retrieve)</td> </tr> <tr> <td>200 OK INVITE(sendrecv)</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>INVITE(sendonly)</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← CPG(hold)</td> </tr> <tr> <td>200 OK INVITE(recvonly)</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>INVITE(sendrecv)</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← CPG(retrieve)</td> </tr> <tr> <td>200 OK INVITE(sendrecv)</td> <td style="text-align: center;">→</td> <td></td> </tr> </tbody> </table>	SIP	MGCF	ISUP	INVITE	→	IAM	180 Ringing	←	ACM	200 OK INVITE	←	ANM	INVITE(sendonly)	→	→ CPG(hold)	200 OK INVITE(recvonly)	←		INVITE(sendrecv)	→	→ CPG(retrieve)	200 OK INVITE(sendrecv)	←		INVITE(sendonly)	←	← CPG(hold)	200 OK INVITE(recvonly)	→		INVITE(sendrecv)	←	← CPG(retrieve)	200 OK INVITE(sendrecv)	→		
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INVITE(sendrecv)	←	← CPG(retrieve)																																				
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TP502002	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10																					
TSS reference:	SIP-ISUP/SS/HOLD/																						
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams.																						
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service. Support the invocation of the service in the alerting state.																						
Test purpose:	Ensure that a party can put the other party on hold in the alerting state. Ensure that the party can retrieve the call previously put on hold. <ul style="list-style-type: none"> • The calling party should be able to put the other party on hold • The calling party should be able to retrieve the other party 																						
SIP Parameter values:	SDP: a=sendonly (put on hold) a=recvonly or omitted (retrieve the call) o= . . <version incremented>																						
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200 OK UPDATE(recvonly)	←																						
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200 OK UPDATE(sendrecv)	←																						

TP502003	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10																		
TSS reference:	SIP-ISUP/SS/HOLD/																			
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams. Support the invocation of the service after the calling user has provided all of the information necessary for processing the call.																			
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service.																			
Test purpose:	Ensure that a party can put the other party on hold after the calling user has provided all of the information necessary for processing the call. Ensure that the party can retrieve the call previously put on hold. <ul style="list-style-type: none"> • The calling party should be able to put the other party on hold • The calling party should be able to retrieve the other party 																			
SIP Parameter values:	SDP: a=sendonly (put on hold) a=recvonly or omitted (retrieve the call) o= . . <version> incremented																			
ISUP Parameter values:	ACM: called party status: no indication CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval Event indicator PROGRESS (retrieve the call)																			
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left;">SIP</th> <th style="text-align: center;">MGCF</th> <th style="text-align: right;">ISUP</th> </tr> </thead> <tbody> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ IAM</td> </tr> <tr> <td>UPDATE(sendonly)</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>200 OK UPDATE(recvonly)</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>UPDATE(sendrecv)</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>200 OK UPDATE(sendrecv)</td> <td style="text-align: center;">←</td> <td></td> </tr> </tbody> </table>	SIP	MGCF	ISUP	INVITE	→	→ IAM	UPDATE(sendonly)	→		200 OK UPDATE(recvonly)	←		UPDATE(sendrecv)	→		200 OK UPDATE(sendrecv)	←		
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TP502004	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10																														
TSS reference:	SIP-ISUP/SS/HOLD/																															
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams.																															
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service.																															
Test purpose:	<p>Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold.</p> <ul style="list-style-type: none"> • The calling party should be able to put the other party on hold • The calling party should be able to retrieve the other party 																															
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TP502005	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10																														
TSS reference:	SIP-ISUP/SS/HOLD/																															
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams. The MGCF sends the update of the media stream in an UPDATE message.																															
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service.																															
Test purpose:	<p>Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold.</p> <ul style="list-style-type: none"> • The called party should be able to put the other party on hold • The called party should be able to retrieve the other party 																															
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200 OK INVITE(recvonly)	→																															
UPDATE(sendrecv)	←	CPG(retrieve)																														
200 OK UPDATE(recvonly)	→																															

TP502006	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10																																				
TSS reference:	SIP-ISUP/SS/HOLD/																																					
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams.																																					
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service.																																					
Test purpose:	<p>Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party in held state can put the remote party put on hold. Ensure that a party can retrieve the call previously put on hold.</p> <ul style="list-style-type: none"> • The calling party should be able to put the other party on hold • The called party should be able to put the other party on hold • The calling party should be able to retrieve the other party • The called party should be able to retrieve the other party 																																					
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TP502007	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10																																				
TSS reference:	SIP-ISUP/SS/HOLD/																																					
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams																																					
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service.																																					
Test purpose:	<p>Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party in held state can put the remote party put on hold. Ensure that a party can retrieve the call previously put on hold.</p> <ul style="list-style-type: none"> • The calling party should be able to put the other party on hold • The called party should be able to put the other party on hold • The called party should be able to retrieve the other party • The calling party should be able to retrieve the other party 																																					
SIP Parameter values:	SDP: a=sendonly or a=inactive (put on hold) a=sendrecv or a=recvonly or omitted (retrieve the call) o= . . . <version incremented>																																					
ISUP Parameter values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)																																					
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP</th> <th style="text-align: center; width: 40%;">MGCF</th> <th style="text-align: right; width: 30%;">ISUP</th> </tr> </thead> <tbody> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→</td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: right;">←</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">←</td> </tr> <tr> <td>INVITE(sendonly)</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ CPG(hold)</td> </tr> <tr> <td>200 OK INVITE(recvonly)</td> <td style="text-align: center;">←</td> <td></td> </tr> <tr> <td>INVITE(inactive)</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← CPG(hold)</td> </tr> <tr> <td>200 OK INVITE(inactive)</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>INVITE(recvonly)</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← CPG(retrieve)</td> </tr> <tr> <td>200 OK INVITE(sendonly)</td> <td style="text-align: center;">→</td> <td></td> </tr> <tr> <td>INVITE(sendrecv)</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ CPG(retrieve)</td> </tr> <tr> <td>200 OK INVITE(sendrecv)</td> <td style="text-align: center;">←</td> <td></td> </tr> </tbody> </table>		SIP	MGCF	ISUP	INVITE	→	→	180 Ringing	←	←	200 OK INVITE	←	←	INVITE(sendonly)	→	→ CPG(hold)	200 OK INVITE(recvonly)	←		INVITE(inactive)	←	← CPG(hold)	200 OK INVITE(inactive)	→		INVITE(recvonly)	←	← CPG(retrieve)	200 OK INVITE(sendonly)	→		INVITE(sendrecv)	→	→ CPG(retrieve)	200 OK INVITE(sendrecv)	←	
SIP	MGCF	ISUP																																				
INVITE	→	→																																				
180 Ringing	←	←																																				
200 OK INVITE	←	←																																				
INVITE(sendonly)	→	→ CPG(hold)																																				
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200 OK INVITE(inactive)	→																																					
INVITE(recvonly)	←	← CPG(retrieve)																																				
200 OK INVITE(sendonly)	→																																					
INVITE(sendrecv)	→	→ CPG(retrieve)																																				
200 OK INVITE(sendrecv)	←																																					

Table 52: Void.

6.3.1.3 Terminal portability (TP)

TP503001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.13																		
TSS reference:	SIP-ISUP/SS/TP/																			
SIP selection criteria:	PICS 8/3																			
ISUP selection criteria:	PICS 5/6																			
Test purpose:	<p>Ensure that the SUT stop the temporarily sending one or more unicast media streams if a SUS message (ISDN subscriber initiated) was received.</p> <p>Ensure that the SUT retrieved the media stream if an RES message (ISDN subscriber initiated) was received.</p>																			
SIP Parameter values:	SDP: a=sendonly or a=inactive (suspended) a=sendrecv or a=recvonly or omitted (resumed)																			
ISUP Parameter values:	SUS: Suspend/Resume indicator ISDN subscriber initiated RES: Suspend/Resume indicator ISDN subscriber initiated																			
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="text-align: left; padding-right: 20px;">SIP</td> <td style="text-align: center; width: 40px;">SUT</td> <td style="text-align: right; width: 40px;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td>→</td> </tr> <tr> <td>180 Ringing</td> <td>←</td> <td>←</td> </tr> <tr> <td>200 OK INVITE</td> <td>←</td> <td>←</td> </tr> <tr> <td> INVITE</td> <td>←</td> <td>←</td> </tr> <tr> <td>INVITE</td> <td>←</td> <td>←</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	→	180 Ringing	←	←	200 OK INVITE	←	←	 INVITE	←	←	INVITE	←	←	
SIP	SUT	ISUP																		
INVITE	→	→																		
180 Ringing	←	←																		
200 OK INVITE	←	←																		
 INVITE	←	←																		
INVITE	←	←																		

TP503002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.13																								
TSS reference:	SIP-ISUP/SS/TP/																									
SIP selection criteria:	PICS 5/6																									
ISUP selection criteria:	PICS 4/14																									
Test purpose:	<p>Ensure that the SUT stop the temporarily sending one or more unicast media streams if a SUS message (ISDN subscriber initiated) was received.</p> <p>Ensure that the connection is cleared after T2 was expired in the PSTN.</p>																									
SIP Parameter values:	SDP: a=sendonly or a=inactive (suspended)																									
ISUP Parameter values:	SUS: Suspend/Resume indicator ISDN subscriber initiated																									
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="text-align: left; padding-right: 20px;">SIP</td> <td style="text-align: center; width: 40px;">SUT</td> <td style="text-align: right; width: 40px;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td>→</td> </tr> <tr> <td>180 Ringing</td> <td>←</td> <td>←</td> </tr> <tr> <td>200 OK INVITE</td> <td>←</td> <td>←</td> </tr> <tr> <td> INVITE</td> <td>←</td> <td>←</td> </tr> <tr> <td></td> <td></td> <td>T2 expiry</td> </tr> <tr> <td>BYE</td> <td>←</td> <td>←</td> </tr> <tr> <td>200 OK BYE</td> <td>→</td> <td>→</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	→	180 Ringing	←	←	200 OK INVITE	←	←	 INVITE	←	←			T2 expiry	BYE	←	←	200 OK BYE	→	→	
SIP	SUT	ISUP																								
INVITE	→	→																								
180 Ringing	←	←																								
200 OK INVITE	←	←																								
 INVITE	←	←																								
		T2 expiry																								
BYE	←	←																								
200 OK BYE	→	→																								

Table 53: Void.

6.3.1.4 Conference calling (CONF)

TP504001	SIP reference: RFC 3261 [6]	NGN reference: ES 283 027 [14], clause 7.4.14																					
TSS reference:	SIP-ISUP/SS/CONF/																						
SIP selection criteria:	PICS 8/2																						
ISUP selection criteria:	PICS 5/10																						
Test purpose:	<p>Ensure that the SUT stop temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value GEN_NOT_VALUE was received due to the CONF supplementary service.</p> <ul style="list-style-type: none"> If the media stream is either in state "sendonly" or "inactive" then: INVITE with the attribute line a_LINE_VA, or omitted attribute line, else: no mapping. 																						
SIP Parameter values:	SDP: a= a_LINE_VA (see table 54) or a line is omitted																						
ISUP Parameter values:	<p>CPG: Generic notification = Conference established</p> <p>CPG: Generic notification = GEN_NOT_VALUE</p>																						
Comments:	<p>SIP SUT ISUP</p> <table> <tbody> <tr> <td>INVITE →</td> <td>→</td> <td>IAM</td> </tr> <tr> <td>180 Ringing ←</td> <td>←</td> <td>ACM</td> </tr> <tr> <td>200 OK INVITE ←</td> <td>←</td> <td>ANM</td> </tr> </tbody> </table> <p>If the media stream is either in state "sendonly" or "inactive"</p> <table> <tbody> <tr> <td>INVITE ←</td> <td>←</td> <td>CPG</td> </tr> <tr> <td>INVITE ←</td> <td>←</td> <td>CPG</td> </tr> <tr> <td>BYE ←</td> <td>←</td> <td>REL</td> </tr> <tr> <td>200 OK BYE →</td> <td>→</td> <td>RLC</td> </tr> </tbody> </table>	INVITE →	→	IAM	180 Ringing ←	←	ACM	200 OK INVITE ←	←	ANM	INVITE ←	←	CPG	INVITE ←	←	CPG	BYE ←	←	REL	200 OK BYE →	→	RLC	
INVITE →	→	IAM																					
180 Ringing ←	←	ACM																					
200 OK INVITE ←	←	ANM																					
INVITE ←	←	CPG																					
INVITE ←	←	CPG																					
BYE ←	←	REL																					
200 OK BYE →	→	RLC																					

Table 54: Void.

TP504003	SIP reference: RFC 3261 [6]	NGN reference: ES 283 027 [14], clause 7.4.14																								
TSS reference:	SIP-ISUP/SS/CONF/																									
SIP selection criteria:	PICS 8/2																									
ISUP selection criteria:	PICS 5/10																									
Test purpose:	Ensure that the SUT stop temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value GEN_NOT_VALUE was received due to the CONF supplementary service. <ul style="list-style-type: none"> If the media stream is either in state "sendonly" or "inactive" then: INVITE with the attribute line a_LINE_VA, or omitted attribute line, else: no mapping. 																									
SIP Parameter values:	SDP: a= a_LINE_VA (see table 55) or a line is omitted																									
ISUP Parameter values:	CPG: Generic notification = Conference established CPG: Generic notification = GEN_NOT_VALUE																									
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP</th> <th style="text-align: center; width: 40%;">SUT</th> <th style="text-align: right; width: 30%;">ISUP</th> </tr> </thead> <tbody> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">IAM</td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: right;">ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">ANM</td> </tr> </tbody> </table> <p style="text-align: center;">If the media stream is either in state "sendonly" or "inactive"</p> <table style="width: 100%; border-collapse: collapse;"> <tbody> <tr> <td style="width: 30%;">INVITE</td> <td style="width: 40%; text-align: center;">←</td> <td style="width: 30%; text-align: right;">CPG</td> </tr> <tr> <td>INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">CPG</td> </tr> <tr> <td>BYE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">REL</td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">RLC</td> </tr> </tbody> </table>	SIP	SUT	ISUP	INVITE	→	IAM	180 Ringing	←	ACM	200 OK INVITE	←	ANM	INVITE	←	CPG	INVITE	←	CPG	BYE	←	REL	200 OK BYE	→	RLC	
SIP	SUT	ISUP																								
INVITE	→	IAM																								
180 Ringing	←	ACM																								
200 OK INVITE	←	ANM																								
INVITE	←	CPG																								
INVITE	←	CPG																								
BYE	←	REL																								
200 OK BYE	→	RLC																								

Table 55: Void.

TP504005	SIP reference: RFC 3261 [6]	NGN reference: ITU-T Rec Q.1912.5 [1], annex B.1 1.7/Q.7344																														
TSS reference:	SIP-ISUP/SS/CONF/																															
SIP selection criteria:	<ul style="list-style-type: none"> NOT PICS 5/10 																															
ISUP selection criteria:																																
Test purpose:	Ensure that the SUT on receipt of a CPG message due to the CONF supplementary service, the Generic notification indicator with the value. No mapping, no disrupting the SIP procedure.																															
SIP Parameter values:	No mapping																															
ISUP Parameter values:	CPG: Generic notification = Conference established CPG: Generic notification = isolated CPG: Generic notification = reattached CPG: Generic notification = Conference disconnected																															
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">SIP</th> <th style="text-align: center; width: 10%;">SUT</th> <th style="text-align: right; width: 60%;">ISUP</th> </tr> </thead> <tbody> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">IAM</td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: right;">ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">ANM</td> </tr> <tr> <td></td> <td style="text-align: center;">↑</td> <td style="text-align: right;">CPG</td> </tr> <tr> <td></td> <td style="text-align: center;">↑</td> <td style="text-align: right;">CPG</td> </tr> <tr> <td></td> <td style="text-align: center;">↑</td> <td style="text-align: right;">CPG</td> </tr> <tr> <td></td> <td style="text-align: center;">↑</td> <td style="text-align: right;">CPG</td> </tr> <tr> <td>BYE</td> <td style="text-align: center;">←</td> <td style="text-align: right;">REL</td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">→</td> <td style="text-align: right;">RLC</td> </tr> </tbody> </table>	SIP	SUT	ISUP	INVITE	→	IAM	180 Ringing	←	ACM	200 OK INVITE	←	ANM		↑	CPG	BYE	←	REL	200 OK BYE	→	RLC										
SIP	SUT	ISUP																														
INVITE	→	IAM																														
180 Ringing	←	ACM																														
200 OK INVITE	←	ANM																														
	↑	CPG																														
	↑	CPG																														
	↑	CPG																														
	↑	CPG																														
BYE	←	REL																														
200 OK BYE	→	RLC																														

6.3.1.5 Three Party service (3PTY)

TP505001	SIP reference: RFC 3261 [6]	NGN reference: ES 283 027 [14], clause 7.4.15																														
TSS reference:	SIP-ISUP/SS/3PTY/																															
SIP selection criteria:	PICS 8/2																															
ISUP selection criteria:	PICS 5/5 AND PICS 5/18																															
Test purpose:	<p>Ensure that the SUT stop the temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value GEN_NOT_VALUE was received due to the 3PTY supplementary service.</p> <ul style="list-style-type: none"> If the media stream is either in state "sendonly" or "inactive" then: INVITE with the attribute line a_LINE_VA, or omitted attribute line, else: no mapping. 																															
SIP Parameter values:	SDP: a=a_LINE_VA (see table 56)																															
ISUP Parameter values:	CPG: notification = remote hold CPG: Generic notification = GEN_NOT_VALUE																															
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="text-align: left; vertical-align: top;">SIP</td> <td style="text-align: center; vertical-align: top;">SUT</td> <td style="text-align: right; vertical-align: top;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td>IAM</td> </tr> <tr> <td>180 Ringing</td> <td>←</td> <td>ACM</td> </tr> <tr> <td>200 OK INVITE</td> <td>←</td> <td>ANM</td> </tr> <tr> <td> </td> <td> </td> <td> </td> </tr> <tr> <td>INVITE</td> <td>←</td> <td>CPG(hold)</td> </tr> <tr> <td>INVITE</td> <td>←</td> <td>CPG</td> </tr> <tr> <td>INVITE</td> <td>←</td> <td>CPG</td> </tr> <tr> <td>BYE</td> <td>←</td> <td>REL</td> </tr> <tr> <td>200 OK BYE</td> <td>→</td> <td>RLC</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	IAM	180 Ringing	←	ACM	200 OK INVITE	←	ANM				INVITE	←	CPG(hold)	INVITE	←	CPG	INVITE	←	CPG	BYE	←	REL	200 OK BYE	→	RLC	
SIP	SUT	ISUP																														
INVITE	→	IAM																														
180 Ringing	←	ACM																														
200 OK INVITE	←	ANM																														
INVITE	←	CPG(hold)																														
INVITE	←	CPG																														
INVITE	←	CPG																														
BYE	←	REL																														
200 OK BYE	→	RLC																														

TP505002	SIP reference: RFC 3261 [6]	NGN reference: ES 283 027 [14], clause 7.4.15																											
TSS reference:	SIP-ISUP/SS/3PTY/																												
SIP selection criteria:	PICS 8/2																												
ISUP selection criteria:	PICS 5/5 AND PICS 5/18																												
Test purpose:	<p>Ensure that the SUT stop the temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value GEN_NOT_VALUE was received due to the 3PTY supplementary service in the ALERTING state.</p> <ul style="list-style-type: none"> If the media stream is either in state "sendonly" or "inactive" then: INVITE with the attribute line a_LINE_VA, or omitted attribute line, else: no mapping. 																												
SIP Parameter values:	SDP: a=a_LINE_VA (see table 56)																												
ISUP Parameter values:	CPG: Generic notification = remote hold CPG: Generic notification = GEN_NOT_VALUE																												
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="text-align: left; vertical-align: top;">SIP</td> <td style="text-align: center; vertical-align: top;">SUT</td> <td style="text-align: right; vertical-align: top;">ISUP</td> </tr> <tr> <td>INVITE</td> <td>→</td> <td>IAM</td> </tr> <tr> <td>180 Ringing</td> <td>←</td> <td>ACM</td> </tr> <tr> <td> </td> <td> </td> <td> </td> </tr> <tr> <td>UPDATE</td> <td>←</td> <td>CPG(hold)</td> </tr> <tr> <td>UPDATE</td> <td>←</td> <td>CPG</td> </tr> <tr> <td>UPDATE</td> <td>←</td> <td>CPG</td> </tr> <tr> <td>BYE</td> <td>←</td> <td>REL</td> </tr> <tr> <td>200 OK BYE</td> <td>→</td> <td>RLC</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	IAM	180 Ringing	←	ACM				UPDATE	←	CPG(hold)	UPDATE	←	CPG	UPDATE	←	CPG	BYE	←	REL	200 OK BYE	→	RLC	
SIP	SUT	ISUP																											
INVITE	→	IAM																											
180 Ringing	←	ACM																											
UPDATE	←	CPG(hold)																											
UPDATE	←	CPG																											
UPDATE	←	CPG																											
BYE	←	REL																											
200 OK BYE	→	RLC																											

Table 56

Values for test purpose TP505001, TP505002		
←INVITE/UPDATE SDP attribute line a_LINE_VA	← CPG Generic notification GEN_NOT_VALUE	
VA_01 sendonly or inactive	Conference established	
VA_02 sendrecv or recvonly or omitted	Conference disconnected	

TP505003	SIP reference: RFC 3261 [6]	NGN reference: ES 283 027 [14], clause 7.4.15 2.7/Q.734	
TSS reference:	SIP-ISUP/SS/3PTY/		
SIP selection criteria:	•		
ISUP selection criteria:	NOT PICS 5/18		
Test purpose:	Ensure that the SUT on receipt of a CPG message due to the 3PTY supplementary service, the Generic notification indicator with the value. No mapping, no disrupting the SIP procedure.		
SIP Parameter values:	No mapping		
ISUP Parameter values:	CPG: Generic notification = Conference established CPG: Generic notification = Conference disconnected		
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← BYE ← 200 OK BYE →	SUT → ← ← ← ← ← ← →	ISUP IAM ACM ANM CPG CPG REL RLC

6.3.1.6 Connected line identification (COL)

TP506001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.2	
TSS reference:	SIP-ISUP/SS/COL/		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	Ensure that the SUT, if a connected number is received in an ANM, does not disrupt the SIP signalling procedure. The connected number is not mapped into any SIP message.		
SIP Parameter values:			
ISUP Parameter values:	ANM: Connected number Parameter		
	SIP INVITE → 180 Ringing ← 200 OK INVITE ← ACK → BYE → 200 OK BYE ←	SUT → ← ← → → ←	ISUP IAM ACM ANM Conversation REL RLC

6.3.1.7 Malicious call identification MCID

TP507001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.4
TSS reference:	SIP-ISUP/SS/MCID/	
SIP selection criteria:	PICS 9/1	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if an IDR is received returns an IRS message. The MCID response indicator is set to "MCID not included". The SIP signalling procedure is not disrupted.	
SIP Parameter values:	No influence	
ISUP Parameter values:	IDR: MCID requested IRS: MCID not included	
Comments:	<p>SIP INVITE → SUT → ISUP 180 Ringing ← SUT ← IAM 200 OK INVITE ← SUT ← IDR Conversation → SUT → IRS Conversation ← SUT ← ACM Conversation ← SUT ← ANM</p> <p>BYE → SUT → REL 200 OK BYE ← SUT ← RLC</p>	

TP507002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.4
TSS reference:	SIP-ISUP/SS/MCID/	
SIP selection criteria:	NOT PICS 9/1	
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if an IDR is received, no IDR is sent. The SIP signalling procedure is not disrupted.	
SIP Parameter values:	No influence	
ISUP Parameter values:	IDR: MCID requested	
Comments:	<p>SIP INVITE → SUT → ISUP 180 Ringing ← SUT ← IAM 200 OK INVITE ← SUT ← IDR</p> <p style="text-align: center;">T39 timeout</p> <p>180 Ringing ← SUT ← ACM 200 OK INVITE ← SUT ← ANM Conversation → SUT → Conversation</p> <p>BYE → SUT → REL 200 OK BYE ← SUT ← RLC</p>	

6.3.1.8 Sub-addressing (SUB)

TP508001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.5
TSS reference:	SIP-ISUP/SS/SUB/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if a Sub address is received in an ATP parameter, the SIP signalling procedure is not disrupted.	
SIP Parameter values:	No mapping into any SIP message	
ISUP Parameter values:	ANM: ATP with a Connected sub-address	
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← Conversation	SUT → Conversation ← ISUP IAM ACM ANM
	BYE → 200 OK BYE ←	→ ← REL RLC

6.3.1.9 Call diversion (CDIV)

TP509001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.6
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if an ACM is received with and call diversion may occur indicator in the optional backward call indicator is set to "call diversion may occur", the SIP signalling procedure is not disrupted (CDa, CFNR).	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM optional backward call indicator	
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← Conversation	SUT → Conversation ← ISUP IAM ACM CPG ANM
	BYE → 200 OK BYE ←	→ ← REL RLC

TP509002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.6
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if a ACM is received called party status indicator "no indication" and containing a Redirection number, call diversion information, redirection number restriction and generic notification set to "Call is diverting" , the SIP signalling procedure is not disrupted (CFU, CFB, Cdi).	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: Redirection number, Call diversion information, Redirection number restriction, Generic notification	
Comments:	SIP INVITE 180 Ringing 200 OK INVITE BYE 200 OK BYE	SUT ISUP IAM ACM ANM REL RLC

TP509003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.6
TSS reference:	SIP-ISUP/SS/CDIV/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if a CPG is received containing a Redirection number, call diversion information, redirection number restriction and generic notification set to "Call is diverting" , the SIP signalling procedure is not disrupted (Cda, CFNR, subsequent redirection).	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: Called party status "Subscriber free" CPG: Redirection number, Call diversion information, Generic notification	
Comments:	SIP INVITE 180 Ringing 200 OK INVITE BYE 200 OK BYE	SUT ISUP IAM ACM CPG ANM REL RLC

TP509004	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.6	
TSS reference:	SIP-ISUP/SS/CDIV/		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	Ensure that the SUT if an ANM is received with redirection number restriction parameter , the SIP signalling procedure is not disrupted.		
SIP Parameter values:	No mapping		
ISUP Parameter values:	ANM: Redirection number restriction		
Comments:	SIP INVITE 180 Ringing 200 OK INVITE Conversation	SUT Conversation	ISUP IAM ACM ANM
	BYE 200 OK BYE	 	REL RLC

6.3.1.10 Call waiting (CW)

TP510001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.9
TSS reference:	SIP-ISUP/SS/CW/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if an ACM with Generic notification parameter = "Call is a waiting call" , the SIP signalling procedure is not disrupted.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: Generic notification parameter = "Call is a waiting call"	
Comments:	SIP INVITE → SUT → ISUP 180 Ringing ← → IAM 200 OK INVITE ← → ACM BYE → Conversation → REL 200 OK BYE ← → ANM	

TP510002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.9																																	
TSS reference:	SIP-ISUP/SS/CW/																																		
SIP selection criteria:																																			
ISUP selection criteria:																																			
Test purpose:	Ensure that the SUT if a CPG with Generic notification parameter = "Call is a waiting call" , the SIP signalling procedure is not disrupted.																																		
SIP Parameter values:	No mapping																																		
ISUP Parameter values:	ACM: Called party status "Subscriber free" CPG: Generic notification parameter = "Call is a waiting call"																																		
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="text-align: left; padding-right: 20px;">SIP</td> <td style="text-align: center; width: 40px;">SUT</td> <td style="text-align: right; width: 40px;">ISUP</td> </tr> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> <tr> <td></td> <td style="text-align: center;">Conversation</td> <td style="text-align: center;">Conversation</td> </tr> <tr> <td>BYE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	→	180 Ringing	←	←	200 OK INVITE	←	←		Conversation	Conversation	BYE	→	→	200 OK BYE	←	←	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="text-align: left; padding-right: 20px;">IAM</td> <td style="text-align: right; width: 40px;">ISUP</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> </tr> <tr> <td>CPG</td> <td style="text-align: center;">←</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> </tr> <tr> <td>REL</td> <td style="text-align: center;">→</td> </tr> <tr> <td>RLC</td> <td style="text-align: center;">←</td> </tr> </table>	IAM	ISUP	ACM	←	CPG	←	ANM	←	REL	→	RLC	←
SIP	SUT	ISUP																																	
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200 OK BYE	←	←																																	
IAM	ISUP																																		
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CPG	←																																		
ANM	←																																		
REL	→																																		
RLC	←																																		

6.3.1.11 User to user signalling (UUS)

TP511001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.3.7.2/Q.737																																						
TSS reference:	SIP-ISUP/SS/UUS/																																							
SIP selection criteria:																																								
ISUP selection criteria:	PICS 11/1 AND PICS 11/2																																							
Test purpose:	Ensure that the SUT if a FAR is received with an user-to-user service 3 request (not essential) after call setup, sent a FRJ to reject the request. The SIP signalling procedure is not disrupted.																																							
SIP Parameter values:																																								
ISUP Parameter values:	FRJ: User-to-user indicator = "Service 3 not provided"																																							
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="text-align: left; padding-right: 20px;">SIP</td> <td style="text-align: center; width: 40px;">SUT</td> <td style="text-align: right; width: 40px;">ISUP</td> </tr> <tr> <td>INVITE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> </tr> <tr> <td>180 Ringing</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> <tr> <td>200 OK INVITE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> <tr> <td></td> <td style="text-align: center;">Conversation</td> <td style="text-align: center;">Conversation</td> </tr> <tr> <td></td> <td style="text-align: center;">Conversation</td> <td style="text-align: center;">Conversation</td> </tr> <tr> <td>BYE</td> <td style="text-align: center;">→</td> <td style="text-align: center;">→</td> </tr> <tr> <td>200 OK BYE</td> <td style="text-align: center;">←</td> <td style="text-align: center;">←</td> </tr> </table>	SIP	SUT	ISUP	INVITE	→	→	180 Ringing	←	←	200 OK INVITE	←	←		Conversation	Conversation		Conversation	Conversation	BYE	→	→	200 OK BYE	←	←	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="text-align: left; padding-right: 20px;">IAM</td> <td style="text-align: right; width: 40px;">ISUP</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> </tr> <tr> <td>FAR</td> <td style="text-align: center;">←</td> </tr> <tr> <td>FRJ</td> <td style="text-align: center;">→</td> </tr> <tr> <td>REL</td> <td style="text-align: center;">→</td> </tr> <tr> <td>RLC</td> <td style="text-align: center;">←</td> </tr> </table>	IAM	ISUP	ACM	←	ANM	←	FAR	←	FRJ	→	REL	→	RLC	←
SIP	SUT	ISUP																																						
INVITE	→	→																																						
180 Ringing	←	←																																						
200 OK INVITE	←	←																																						
	Conversation	Conversation																																						
	Conversation	Conversation																																						
BYE	→	→																																						
200 OK BYE	←	←																																						
IAM	ISUP																																							
ACM	←																																							
ANM	←																																							
FAR	←																																							
FRJ	→																																							
REL	→																																							
RLC	←																																							

TP511002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.3.5.2.5.2.1/Q.737
TSS reference:	SIP-ISUP/SS/UUS/	
SIP selection criteria:		
ISUP selection criteria:	NO PICS 11/2	
Test purpose:	Ensure that the SUT if a FAR is received with an user-to-user service 3 request (not essential) after call setup, the SIP signalling procedure is not disrupted.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← Conversation → Conversation ← BYE → 200 OK BYE ←	SUT → ← ← Conversation → Conversation ← Conversation → Conversation ← ISUP IAM ACM ANM FAR REL RLC

6.3.1.12 Explicit call transfer (ECT)

TP512001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.8
TSS reference:	SIP-ISUP/SS/ECT/	
SIP selection criteria:		
ISUP selection criteria:	PICS 12/1	
Test purpose:	Ensure that the SUT if a LOP(request) is received returns a LOP(response) with the indication "insufficient information" continue without disrupting the SIP signalling procedure. Ensure that the SUT if a FAC is received continue without disrupting the SIP signalling procedure.	
SIP Parameter values:		
ISUP Parameter values:	LOP: Response "insufficient information"	
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← Conversation → BYE → 200 OK BYE ←	SUT → ← ← Conversation → ISUP IAM ACM ANM LOP LOP FAC Conversation → Conversation ← REL RLC

TP512002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.8
TSS reference:	SIP-ISUP/SS/ECT/	
SIP selection criteria:		
ISUP selection criteria:	NO PICS 12/1	
Test purpose:	Ensure that the SUT if a LOP(request) is received continue without disrupting the SIP signalling procedure. Ensure that the SUT if a FAC is received continue without disrupting the SIP signalling procedure.	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<p>SIP INVITE → 180 Ringing ← 200 OK INVITE ←</p> <p style="text-align: center;">Conversation</p> <p>BYE → 200 OK BYE ←</p>	<p>SUT → Conversation ← Conversation ← Conversation ← Conversation → Conversation →</p> <p style="text-align: center;">Conversation</p> <p>ISUP IAM ACM ANM LOP FAC REL RLC</p>

6.3.1.13 Completion of Call to Busy Subscriber (CCBS)

TP513001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.11
TSS reference:	SIP-ISUP/SS/CCBS/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if a REL is received contained a Diagnostic field and the CCBS indicator is coded as CCBS possible: <ul style="list-style-type: none"> continue without disrupting the SIP signalling procedure. 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	<p>SIP INVITE → 486 Busy Here ← ACK →</p>	<p>SUT → ← ← →</p> <p style="text-align: center;">REL</p> <p>ISUP IAM REL RLC</p>

6.3.1.14 Completion of Calls on No reply (CCNR)

TP514001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.12	
TSS reference:	SIP-ISUP/SS/CCNR/		
SIP selection criteria:			
ISUP selection criteria:			
Test purpose:	Ensure that the SUT if a ACM is received and a CCNR Possible Indicator is included: <ul style="list-style-type: none"> • continue without disrupting the SIP signalling procedure. 		
SIP Parameter values:			
ISUP Parameter values:			
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← Conversation	SUT → ← ← Conversation	ISUP IAM ACM ANM
	BYE → 200 OK BYE ←	→ ←	REL RLC

6.3.1.15 Anonymous Call Rejection (ACR)

TP515001	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [14], clause 7.4.23	
TSS reference:	SIP-ISUP/SS/ACR/		
SIP selection criteria:			
ISUP selection criteria:	PICS 1/9		
Test purpose:	Ensure that the SUT, if a destination user has subscribed the ACR supplementary service: <ul style="list-style-type: none"> • the call attempt is rejected with a REL cause value 24 "call rejected due to ACR supplementary service". 		
SIP Parameter values:	INVITE: Privacy-header = "id" 603 Decline: Reason header field Reason: ITU-T Rec Q.850 [5];cause=24		
ISUP Parameter values:	REL: Cause value: 24 "call rejected due to ACR supplementary service"		
Comments:	SIP INVITE → 603 Decline ← ACK →	SUT → ← →	ISUP IAM REL RLC

TP515002	SIP reference: RFC 3261 [6]	ISUP reference: ES 283 027 [14], clause 7.4.23
TSS reference:	SIP-ISUP/SS/ACR/	
SIP selection criteria:		
ISUP selection criteria:	PICS 1/9 AND PICS 6/12	
Test purpose:	Ensure that the SUT if a destination user has subscribed the ACR supplementary service: <ul style="list-style-type: none"> • the call attempt is successful. 	
SIP Parameter values:	INVITE: No P-Asserted-Identity header field and no Privacy header field present	
ISUP Parameter values:	IAM: Calling party number Address presentation restriction is set to "Presentation restricted by the network"	
Comments:	SIP INVITE → 180 Ringing ← 200 OK INVITE ← Conversation BYE → 200 OK BYE ←	SUT → → ← → ← Conversation → ← REL RLC

6.3.2 Interworking from ISUP to SIP (Outgoing Call)

6.3.2.1 Calling Line Identification (CLI)

TP601001	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that when the SUT has received an IAM message whereby Calling Party Number parameter and the Generic Number are not applicable: <ul style="list-style-type: none"> • Sends an INVITE message without the "P-Asserted-Identity header field", a "From header field" set to unavailable@hostportion and without a "Privacy Header field". 	
SIP Parameter values:		
ISUP Parameter values:		
Comments:	ISUP/BICC IAM	SUT → SIP INVITE

TP601002	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	PICS 4/13	
ISUP selection criteria:		
Test purpose:	Ensure that when the SUT has received an IAM message whereby Calling Party Number parameter is not applicable and the Generic Number is applicable whereby the address presentation restriction parameter is set to "presentation allowed" and the Nature of Address Indicator is set to NoAS_VALUE: <ul style="list-style-type: none"> • Sends an INVITE message without the "P-Asserted-Identity header field", a "From header field" and no a "Privacy Header field". 	
SIP Parameter values:	P-Asserted-Identity header field: not included: From header field: Display-name (optional) and addr-spec: Addr-spec: Addr_SPEC_ID Display-name: display-name is derived from the Generic number (AcgPN) Privacy header: is not included	
ISUP Parameter values:	Generic Number: "additional calling party number" Nature of Address Indicator: NoAS_VALUE	
Comments:	ISUP/BICC IAM	SUT → SIP INVITE

Table 57

Values for test purpose TP601002;		
	ISUP Parameter values:	SIP Parameter values:
VA_01	IAM NoAS_VALUE: " <i>national (significant) number</i> "(NDC+SN)	INVITE FHf_Addr_SPEC_ID: CC (of the country where the IWU is located) is added to the Generic Number Address Signals and then mapped to user portion of URI scheme
VA_02	IAM NoAS_VALUE: " <i>international number</i> " ("+"CC+NDC+SN)	INVITE FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used.

TP601003	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	NOT PICS 4/13	
ISUP selection criteria:		
Test purpose:	Ensure that when the SUT has received an IAM message whereby Calling Party Number parameter is not applicable and the Generic Number is applicable whereby the address presentation restriction parameter is set to "presentation allowed" and the Nature of Address Indicator is set to NoAS_VALUE: <ul style="list-style-type: none"> • Sends an INVITE message without the "P-Asserted-Identity header field", a "From header field" and no a "Privacy Header field". 	
SIP Parameter values:	P-Asserted-Identity header field: not included: From header field: Display-name (optional) and addr-spec: Addr-spec: Addr_SPEC_ID Display-name: not supported Privacy header: is not included.	
ISUP Parameter values:	Generic Number: "additional calling party number" Nature of Address Indicator: NoAS_VALUE	
Comments:	ISUP/BICC IAM	SUT → SIP INVITE

Table 58

Values for test purpose TP601003;		
	ISUP Parameter values:	SIP Parameter values:
VA_01	IAM NoAS_VALUE: " <i>national (significant) number</i> " (NDC+SN)	INVITE FHf_Addr_SPEC_ID: CC (of the country where the IWU is located) is added to the Generic Number Address Signals and then mapped to user portion of URI scheme
VA_02	IAM NoAS_VALUE: " <i>international number</i> " ("+"CC+NDC+SN)	INVITE FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used.

TP601004	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	PICS 4/13 AND PICS 4/20	
ISUP selection criteria:		
Test purpose:	<p>Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation allowed and the Generic Number is not applicable:</p> <ul style="list-style-type: none"> Sends an INVITE message with the "P-Asserted-Identity header field" where the "addr-spec" is set to PAIh_Addr_SPEC_ID, a "From header field" where the "addr-spec" is set to FHf_Addr_SPEC_ID without "Privacy Header field" or "id" is not included. 	
SIP Parameter values:	<p>P-Asserted-Identity header field:</p> <p>Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals)</p> <p>Display-name: display-name is mapped from CgPN Address Signals</p> <p>From header field: Display-name (optional) and addr-spec:</p> <p>Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals)</p> <p>Display-name: Display-name: display-name is mapped from CgPN Address Signals</p> <p>Privacy header: is not included or if included, "id" is not included</p>	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM	SUT → SIP INVITE

Table 59

Values for test purpose TP601004;		
	ISUP Parameter values:	SIP Parameter values:
VA_01	IAM NoAS_VALUE: " <i>national (significant number</i> "(NDC+SN)	INVITE PAIh_Addr_SPEC_ID = FHf_Addr_SPEC_ID: Add CC (of the country where the IWU is located) to CgPN Signals then map to user portion of URI scheme used
VA_02	IAM NoAS_VALUE: " <i>international number</i> " ("+"CC+NDC+SN)	INVITE PAIh_Addr_SPEC_ID= FHf_Addr_SPEC_ID: the complete to CgPN Signals is mapped to the user portion of URI scheme.

TP601005	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	NOT PICS 4/13 AND PICS 4/20	
ISUP selection criteria:		
Test purpose:	<p>Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation allowed and the Generic Number is not applicable:</p> <ul style="list-style-type: none"> • Sends an INVITE message with the "P-Asserted-Identity header field" where the "addr-spec" is set to PAIh_Addr_SPEC_ID; • a "From header field" where the "addr-spec" is set to FHf_Addr_SPEC_ID; • without "Privacy Header field" or "id" is not supported. 	
SIP Parameter values:	<p>P-Asserted-Identity header field:</p> <p>Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals)</p> <p>Display-name: display-name is mapped from CgPN Address Signals</p> <p>From header field: Display-name (optional) and addr-spec:</p> <p>Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals)</p> <p>Display-name: not supported</p> <p>Privacy header: is not included or if included, "id" is not included.</p>	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM	SUT → SIP INVITE

Table 60

Values for test purpose TP601005		
	ISUP Parameter values:	SIP Parameter values:
VA_01	IAM NoAS_VALUE: " <i>national (significant) number</i> "(NDC+SN)	INVITE PAIh_Addr_SPEC_ID = FHf_Addr_SPEC_ID: Add CC (of the country where the IWU is located) to CgPN Signals then map to user portion of URI scheme used
VA_02	IAM NoAS_VALUE: " <i>international number</i> " ("+"CC+NDC+SN)	INVITE PAIh_Addr_SPEC_ID= FHf_Addr_SPEC_ID: the complete to CgPN Signals is mapped to the user portion of URI scheme.

TP601006	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	PICS 4/13 AND PICS 4/20	
ISUP selection criteria:		
Test purpose:	Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation restricted and the Generic Number is not applicable: <ul style="list-style-type: none"> Sends an INVITE message with the "P-Asserted-Identity header field" where the "addr-spec" is set to PAIh_Addr_SPEC_ID, a "From header field" where the "addr-spec" is set to FHf_Addr_SPEC_ID and with "Privacy Header field". 	
SIP Parameter values:	P-Asserted-Identity header field: Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) Display-name: display-name is mapped from CgPN Address Signals From header field: Display-name (optional) and addr-spec: Addr-spec: Anonymous@Anonymous.invalid Display-name: Anonymous Privacy header: "id".	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM	SUT → SIP → INVITE

Table 61

Values for test purpose TP601006		
	ISUP Parameter values:	SIP Parameter values:
VA_01	IAM NoAS_VALUE: " <i>national (significant) number</i> "(NDC+SN)	INVITE PAIh_Addr_SPEC_ID = FHf_Addr_SPEC_ID: CC (of the country where the IWU is located) is added to the CgPN Signals and then mapped to user portion of URI scheme used
VA_02	IAM NoAS_VALUE: " <i>international number</i> " ("+"CC+NDC+SN)	INVITE PAIh_Addr_SPEC_ID= FHf_Addr_SPEC_ID: the complete to CgPN Signals is mapped to the user portion of URI scheme.

TP601007	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	NOT PICS 4/13 AND PICS 4/20	
ISUP selection criteria:		
Test purpose:	Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation restricted and the Generic Number is not applicable: <ul style="list-style-type: none"> Sends an INVITE message with the "P-Asserted-Identity header field" where the "addr-spec" is set to PAIh_Addr_SPEC_ID, a "From header field" where the "addr-spec" is set to FHf_Addr_SPEC_ID and with "Privacy Header field". 	
SIP Parameter values:	P-Asserted-Identity header field: Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) Display-name: display-name is mapped from CgPN Address Signals From header field: Display-name (optional) and addr-spec: Addr-spec: Anonymous@Anonymous.invalid Display-name: not supported Privacy header: "id".	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM	SUT → SIP → INVITE

Table 62

Values for test purpose TP601007		
	ISUP Parameter values:	SIP Parameter values:
VA_01	IAM NoAS_VALUE: " <i>national (significant) number</i> "(NDC+SN)	INVITE PAIh_Addr_SPEC_ID = FHf_Addr_SPEC_ID: CC (of the country where the IWU is located) is added to the CgPN Signals and then mapped to user portion of URI scheme used
VA_02	IAM NoAS_VALUE: " <i>international number</i> " ("+"CC+NDC+SN)	INVITE PAIh_Addr_SPEC_ID= FHf_Addr_SPEC_ID: the complete to CgPN Signals is mapped to the user portion of URI scheme.

TP601008	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	PICS 4/13 AND NOT PICS 4/20	
ISUP selection criteria:		
Test purpose:	<p>Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation allowed and the Generic Number is applicable:</p> <ul style="list-style-type: none"> Sends an INVITE message with the "P-Asserted-Identity header field", where the "addr-spec" is set to PAIh_Addr_SPEC_ID "From header field" where the "addr-spec" is set to FH_Addr_SPEC_ID and without "Privacy Header field" or "id" is not included. 	
SIP Parameter values:	<p>P-Asserted-Identity header field:</p> <p>Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals)</p> <p>Display-name: not supported</p> <p>From header field: Display-name (optional) and addr-spec:</p> <p>Addr-spec: FH_Addr_SPEC_ID (Derived from Generic Number parameter Address Signals (AcgPN))</p> <p>Display-name: display-name is mapped from ACgPN Address Signals</p> <p>Privacy header: is not included or if included, "id" is not included.</p>	
ISUP Parameter hbvalues:	<p>Generic Number: "additional calling party number"</p> <p>Nature of Address Indicator: CP_NoAS_VALUE</p> <p>APRI: presentation allowed</p>	
Comments:	ISUP/BICC IAM	SUT → SIP INVITE

Table 62a

Values for test purpose TP601008			
Test purposes	ISUP Parameter values:	SIP Parameter values:	
VA_01	IAM NoAS_VALUE: " <i>national (significant) number</i> "(NDC+SN)	INVITE FHf_Addr_SPEC_ID: Add CC (of the country where the IWU is located) to CgPN Signals then map to user portion of URI scheme used	INVITE PAIh_Addr_SPEC_ID: Add CC (of the country where the IWU is located) to CgPN Signals then map to user portion of URI scheme used
VA_02	IAM NoAS_VALUE: " <i>international number</i> " ("+"CC+NDC+SN)	INVITE FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme.	INVITE PAIh_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used.

TP601009	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	NOT PICS 4/13 AND NOT PICS 4/20	
ISUP selection criteria:		
Test purpose:	<p>Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation allowed and the Generic Number is applicable:</p> <ul style="list-style-type: none"> Sends an INVITE message with the "P-Asserted-Identity header field", where the "addr-spec" is set to PAIh_Addr_SPEC_ID "From header field" where the "addr-spec" is set to FH_Addr_SPEC_ID and without "Privacy Header field" or "id" is not included. 	
SIP Parameter values:	<p>P-Asserted-Identity header field:</p> <p>Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals)</p> <p>Display-name: not supported</p> <p>From header field: Display-name (optional) and addr-spec:</p> <p>Addr-spec: FH_Addr_SPEC_ID (Derived from Generic Number parameter Address Signals (AcgPN))</p> <p>Display-name: not supported</p> <p>Privacy header: is not included or if included, "id" is not included.</p>	
ISUP Parameter hbvalues:	<p>Generic Number: "additional calling party number"</p> <p>Nature of Address Indicator: CP_NoAS_VALUE</p> <p>APRI: presentation restricted</p>	
Comments:	ISUP/BICC IAM	SUT → SIP → INVITE

Table 63

Values for test purpose TP601009			
	ISUP Parameter values:	SIP Parameter values:	
VA_01	IAM NoAS_VALUE: " <i>national (significant) number</i> "(NDC+SN)	INVITE FHf_Addr_SPEC_ID: Add CC (of the country where the IWU is located) to CgPN Signals then map to user portion of URI scheme used	INVITE PAIh_Addr_SPEC_ID: Add CC (of the country where the IWU is located) to CgPN Signals then map to user portion of URI scheme used
VA_02	IAM NoAS_VALUE: " <i>international number</i> " ("+"CC+NDC+SN)	INVITE FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme.	INVITE PAIh_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used.

TP601010	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	PICS 4/13 AND PICS NOT 4/20	
ISUP selection criteria:		
Test purpose:	Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation restricted and the Generic Number is applicable: <ul style="list-style-type: none"> Sends an INVITE message with the "P-Asserted-Identity header field", where the "addr-spec" is set to PAIh_Addr_SPEC_ID "From header field" where the "addr-spec" is set to FH_Addr_SPEC_ID and with "Privacy Header field". 	
SIP Parameter values:	P-Asserted-Identity header field: Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) Display-name: not supported From header field: Display-name (optional) and addr-spec: Addr-spec: Anonymous@Anonymous.invalid Display-name: Anonymous Privacy header: "id".	
ISUP Parameter values:	Generic Number: "additional calling party number" Nature of Address Indicator: NoAS_VALUE APRI: presentation restricted	
Comments:	ISUP/BICC IAM	SUT → SIP INVITE

Table 64

Values for test purpose TP601010			
	ISUP Parameter values:	SIP Parameter values:	
VA_01	IAM NoAS_VALUE: " <i>national (significant) number</i> "(NDC+SN)	INVITE FHf_Addr_SPEC_ID: Add CC (of the country where the IWU is located) to CgPN Signals then map to user portion of URI scheme used	INVITE PAIh_Addr_SPEC_ID: Add CC (of the country where the IWU is located) to CgPN Signals then map to user portion of URI scheme used
VA_02	IAM NoAS_VALUE: " <i>international number</i> " ("+"CC+NDC+SN)	INVITE FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme.	INVITE PAIh_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used.

TP601011	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	NOT PICS 4/13 AND PICS NOT 4/25	
ISUP selection criteria:		
Test purpose:	Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation restricted and the Generic Number is applicable: <ul style="list-style-type: none"> Sends an INVITE message with the "P-Asserted-Identity header field", where the "addr-spec" is set to PAIh_Addr_SPEC_ID "From header field" where the "addr-spec" is set to FH_Addr_SPEC_ID and with "Privacy Header field". 	
SIP Parameter values:	P-Asserted-Identity header field: Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) Display-name: not supported From header field: Display-name (optional) and addr-spec: Addr-spec: Anonymous@Anonymous .invalid Display-name: not supported Privacy header: "id".	
ISUP Parameter values:	Generic Number: "additional calling party number" Nature of Address Indicator: NoAS_VALUE APRI: presentation restricted	
Comments:	ISUP/BICC IAM	SUT → SIP → INVITE

TP601012	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	PICS 4/13 AND PICS 4/20	
ISUP selection criteria:		
Test purpose:	Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation allowed and the Generic Number is applicable . <ul style="list-style-type: none"> Sends an INVITE message with the "P-Asserted-Identity header field", where the "addr-spec" is set to PAIh_Addr_SPEC_ID "From header field" where the "addr-spec" is set to FH_Addr_SPEC_ID and without "Privacy Header field" or "id" is not included. 	
SIP Parameter values:	P-Asserted-Identity header field: Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) Display-name: display-name is mapped from CgPN Address Signals From header field: Display-name (optional) and addr-spec Addr-spec: FH_Addr_SPEC_ID (Derived from Generic Number parameter Address Signals (AcgPN)) Display-name: display-name is mapped from ACgPN Address Signals Privacy header: is not included or if included, "id" is not included	
ISUP Parameter values:	Generic Number: "additional calling party number" Nature of Address Indicator: CP_NoAS_VALUE APRI: presentation allowed	
Comments:	ISUP/BICC IAM	SUT → SIP → INVITE

TP601013	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	PICS 4/13 AND PICS 4/20	
ISUP selection criteria:		
Test purpose:	<p>Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation restricted and the Generic Number is applicable.</p> <p>Sends an INVITE message with the "P-Asserted-Identity header field", where the "addr-spec" is set to PAIh_Addr_SPEC_ID "From header field" where the "addr-spec" is set to FH_Addr_SPEC_ID and with "Privacy Header field =id".</p>	
SIP Parameter values:	<p>P-Asserted-Identity header field:</p> <p>Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) Display-name: display-name is mapped from CgPN Address Signals</p> <p>From header field: Display-name (optional) and addr-spec Addr-spec: FH_Addr_SPEC_ID (Derived from Generic Number parameter Address Signals (AcgPN)) Display-name: Anonymous</p> <p>Privacy header: "id"</p>	
ISUP Parameter values:	Generic Number: " <i>additional calling party number</i> " Nature of Address Indicator: CP_NoAS_VALUE APRI: presentation restricted	
Comments:	ISUP/BICC IAM	SUT → INVITE

TP601014	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	NOT PICS 4/13 AND PICS 4/20	
ISUP selection criteria:		
Test purpose:	<p>Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation allowed and the Generic Number is applicable.</p> <ul style="list-style-type: none"> Sends an INVITE message with the "P-Asserted-Identity header field", where the "addr-spec" is set to PAIh_Addr_SPEC_ID; "From header field" where the "addr-spec" is set to FH_Addr_SPEC_ID and without "Privacy Header field" or "id" is not included. 	
SIP Parameter values:	<p>P-Asserted-Identity header field:</p> <p>Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) Display-name: display-name is mapped from CgPN Address Signals</p> <p>From header field: Display-name (optional) and addr-spec Addr-spec: FH_Addr_SPEC_ID (Derived from Generic Number parameter Address Signals (AcgPN)) Display-name: not supported</p> <p>Privacy header: is not included or if included, "id" is not included</p>	
ISUP Parameter values:	Generic Number: " <i>additional calling party number</i> " Nature of Address Indicator: CP_NoAS_VALUE APRI: presentation allowed	
Comments:	ISUP/BICC IAM	SUT → INVITE

TP601015	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 clause 7.1.3 [2]
TSS reference:	ISUP-SIP/SS/CLI/	
SIP selection criteria:	NOT PICS 4/13 AND PICS 4/20	
ISUP selection criteria:		
Test purpose:	Ensure that when the SUT has received an IAM message, the Calling Party Number is applicable whereby the Nature of Address Indicator is set to NoAS_VALUE the APRI is set to presentation restricted and the Generic Number is applicable . Sends an INVITE message with the "P-Asserted-Identity header field", where the "addr-spec" is set to PAIh_Addr_SPEC_ID; "From header field" where the "addr-spec" is set to FH_Addr_SPEC_ID and with "Privacy Header field =id".	
SIP Parameter values:	P-Asserted-Identity header field: Addr-spec: PAIh_Addr_SPEC_ID (Derived from Calling Party Number parameter Address Signals) Display-name: display-name is mapped from CgPN Address Signals From header field: Display-name (optional) and addr-spec Addr-spec: FH_Addr_SPEC_ID (Derived from Generic Number parameter Address Signals (AcgPN)) Display-name: not supported Privacy header: "id"	
ISUP Parameter values:	Generic Number: " <i>additional calling party number</i> " Nature of Address Indicator: CP_NoAS_VALUE APRI: presentation restricted	
Comments:	ISUP/BICC IAM	SUT → INVITE

Table 65

Values for test purpose TP601011; TP601012; TP601013, TP601014 and TP601015			
Test purposes	ISUP Parameter values:	SIP Parameter values:	
VA_01	IAM NoAS_VALUE: " <i>national (significant) number</i> "(NDC+SN)	INVITE FHf_Addr_SPEC_ID: Add CC (of the country where the IWU is located) to CgPN Signals then map to user portion of URI scheme used	INVITE PAIh_Addr_SPEC_ID: Add CC (of the country where the IWU is located) to CgPN Signals then map to user portion of URI scheme used
VA_02	IAM NoAS_VALUE: " <i>international number</i> " ("+"CC+NDC+SN)	INVITE FHf_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme.	INVITE PAIh_Addr_SPEC_ID: the complete GenericNumber Address Signals is mapped to the user portion of URI scheme used.

6.3.2.2 Call Hold (HOLD)

TP602001	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10																								
TSS reference:	ISUP-SIP/SS/HOLD/																									
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams																									
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service																									
Test purpose:	Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold. <ul style="list-style-type: none"> • The calling party should be able to put the other party on hold • The calling party should be able to retrieve the other party • The called party should be able to put the other party on hold • The called party should be able to retrieve the other party 																									
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>																									
ISUP Parameter values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)																									
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">ISUP/BICC</th> <th style="text-align: center; width: 10%;">MGCF</th> <th style="text-align: left; width: 60%;">SIP</th> </tr> </thead> <tbody> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td>→ INVITE</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td>← 180 Ringing</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> <td>← 200 OK INVITE</td> </tr> <tr> <td>CPG(hold)</td> <td style="text-align: center;">→</td> <td>→ INVITE(sendonly) ← 200 OK INVITE(recvonly)</td> </tr> <tr> <td>CPG(retrieve)</td> <td style="text-align: center;">→</td> <td>→ INVITE(sendrecv) ← 200 OK INVITE(sendrecv)</td> </tr> <tr> <td>CPG(hold)</td> <td style="text-align: center;">←</td> <td>← INVITE(sendonly) → 200 OK INVITE(recvonly)</td> </tr> <tr> <td>CPG(retrieve)</td> <td style="text-align: center;">←</td> <td>← INVITE(sendrecv) → 200 OK INVITE(sendrecv)</td> </tr> </tbody> </table>	ISUP/BICC	MGCF	SIP	IAM	→	→ INVITE	ACM	←	← 180 Ringing	ANM	←	← 200 OK INVITE	CPG(hold)	→	→ INVITE(sendonly) ← 200 OK INVITE(recvonly)	CPG(retrieve)	→	→ INVITE(sendrecv) ← 200 OK INVITE(sendrecv)	CPG(hold)	←	← INVITE(sendonly) → 200 OK INVITE(recvonly)	CPG(retrieve)	←	← INVITE(sendrecv) → 200 OK INVITE(sendrecv)	
ISUP/BICC	MGCF	SIP																								
IAM	→	→ INVITE																								
ACM	←	← 180 Ringing																								
ANM	←	← 200 OK INVITE																								
CPG(hold)	→	→ INVITE(sendonly) ← 200 OK INVITE(recvonly)																								
CPG(retrieve)	→	→ INVITE(sendrecv) ← 200 OK INVITE(sendrecv)																								
CPG(hold)	←	← INVITE(sendonly) → 200 OK INVITE(recvonly)																								
CPG(retrieve)	←	← INVITE(sendrecv) → 200 OK INVITE(sendrecv)																								

Table 66: Void

TP602002	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10																												
TSS reference:	ISUP-SIP/SS/HOLD/																													
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams Support the invocation of the service in the alerting state																													
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service																													
Test purpose:	Ensure that a party can put the other party on hold in the alerting state. Ensure that the party can retrieve the call previously put on hold. <ul style="list-style-type: none"> • The calling party should be able to put the other party on hold • The calling party should be able to retrieve the other party 																													
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . . <version incremented>																													
ISUP Parameter values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)																													
Comments:	<table> <thead> <tr> <th></th> <th>ISUP/BICC</th> <th>MGCF</th> <th>SIP</th> </tr> </thead> <tbody> <tr> <td>IAM</td> <td>→</td> <td>→</td> <td>INVITE</td> </tr> <tr> <td>ACM</td> <td>←</td> <td>←</td> <td>180 Ringing</td> </tr> <tr> <td>CPG(hold)</td> <td>→</td> <td>→</td> <td>UPDATE(sendonly)</td> </tr> <tr> <td></td> <td></td> <td>←</td> <td>200 OK UPDATE(recevonly)</td> </tr> <tr> <td>CPG(retrieve)</td> <td>→</td> <td>→</td> <td>UPDATE(sendrecv)</td> </tr> <tr> <td></td> <td></td> <td>←</td> <td>200 OK UPDATE(sendrecv)</td> </tr> </tbody> </table>		ISUP/BICC	MGCF	SIP	IAM	→	→	INVITE	ACM	←	←	180 Ringing	CPG(hold)	→	→	UPDATE(sendonly)			←	200 OK UPDATE(recevonly)	CPG(retrieve)	→	→	UPDATE(sendrecv)			←	200 OK UPDATE(sendrecv)	
	ISUP/BICC	MGCF	SIP																											
IAM	→	→	INVITE																											
ACM	←	←	180 Ringing																											
CPG(hold)	→	→	UPDATE(sendonly)																											
		←	200 OK UPDATE(recevonly)																											
CPG(retrieve)	→	→	UPDATE(sendrecv)																											
		←	200 OK UPDATE(sendrecv)																											

TP602003	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10
TSS reference:	ISUP-SIP/SS/HOLD/	
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams	
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service	
Test purpose:	Ensure that a party can put the other party on hold after the calling user has provided all of the information necessary for processing the call. Ensure that the party can retrieve the call previously put on hold. <ul style="list-style-type: none"> • The calling party should be able to put the other party on hold • The calling party should be able to retrieve the other party 	
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>	
ISUP Parameter values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)	
Comments:	ISUP/BICC IAM → ACM ← ANM ← CPG(hold) ← CPG(retrieve) ←	MGCF SIP → INVITE ← 180 Ringing ← 200 OK INVITE ← UPDATE(sendonly) → 200 OK UPDATE(recevonly) ← UPDATE(sendrecv) → 200 OK UPDATE(sendrecv)

TP602004	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10																		
TSS reference:	ISUP-SIP/SS/HOLD/																			
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams The MGCF sends the update of the media stream in an UPDATE message																			
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service																			
Test purpose:	<p>Ensure that a party can put the other party on hold in the alerting state. Ensure that the party can retrieve the call previously put on hold.</p> <ul style="list-style-type: none"> • The calling party should be able to put the other party on hold • The calling party should be able to retrieve the other party 																			
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>																			
ISUP Parameter values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)																			
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left;">ISUP/BICC</th> <th style="text-align: center;">MGCF</th> <th style="text-align: right;">SIP</th> </tr> </thead> <tbody> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ INVITE</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← 180 Ringing</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← 200 OK INVITE</td> </tr> <tr> <td>CPG(hold)</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ UPDATE(sendonly) ← 200 OK UPDATE(recvonly)</td> </tr> <tr> <td>CPG(retrieve)</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ UPDATE(sendrecv) ← 200 OK UPDATE(sendrecv)</td> </tr> </tbody> </table>		ISUP/BICC	MGCF	SIP	IAM	→	→ INVITE	ACM	←	← 180 Ringing	ANM	←	← 200 OK INVITE	CPG(hold)	→	→ UPDATE(sendonly) ← 200 OK UPDATE(recvonly)	CPG(retrieve)	→	→ UPDATE(sendrecv) ← 200 OK UPDATE(sendrecv)
ISUP/BICC	MGCF	SIP																		
IAM	→	→ INVITE																		
ACM	←	← 180 Ringing																		
ANM	←	← 200 OK INVITE																		
CPG(hold)	→	→ UPDATE(sendonly) ← 200 OK UPDATE(recvonly)																		
CPG(retrieve)	→	→ UPDATE(sendrecv) ← 200 OK UPDATE(sendrecv)																		

TP602005	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10																								
TSS reference:	ISUP-SIP/SS/HOLD/																									
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams																									
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service																									
Test purpose:	<p>Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold.</p> <ul style="list-style-type: none"> • The calling party should be able to put the other party on hold • The called party should be able to put the other party on hold • The calling party should be able to retrieve the other party • The called party should be able to retrieve the other party 																									
SIP Parameter values:	SDP: a=sendonly (put on hold) a=sendrecv or omitted (retrieve the call) o= . . <version incremented>																									
ISUP Parameter values:	CPG: Generic notification: remote hold Event indicator PROGRESS (put on hold) Generic notification: remote retrieval event indicator PROGRESS (retrieve the call)																									
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left;">ISUP/BICC</th> <th style="text-align: center;">MGCF</th> <th style="text-align: right;">SIP</th> </tr> </thead> <tbody> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ INVITE</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← 180 Ringing</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← 200 OK INVITE</td> </tr> <tr> <td>CPG(hold)</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ INVITE(sendonly) ← 200 OK INVITE(recvonly)</td> </tr> <tr> <td>CPG(hold)</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← INVITE(inactive) → 200 OK INVITE(inactive)</td> </tr> <tr> <td>CPG(retrieve)</td> <td style="text-align: center;">→</td> <td style="text-align: right;">→ INVITE(recvonly) ← 200 OK INVITE(sendonly)</td> </tr> <tr> <td>CPG(retrieve)</td> <td style="text-align: center;">←</td> <td style="text-align: right;">← INVITE(sendrecv) → 200 OK INVITE(sendrecv)</td> </tr> </tbody> </table>		ISUP/BICC	MGCF	SIP	IAM	→	→ INVITE	ACM	←	← 180 Ringing	ANM	←	← 200 OK INVITE	CPG(hold)	→	→ INVITE(sendonly) ← 200 OK INVITE(recvonly)	CPG(hold)	←	← INVITE(inactive) → 200 OK INVITE(inactive)	CPG(retrieve)	→	→ INVITE(recvonly) ← 200 OK INVITE(sendonly)	CPG(retrieve)	←	← INVITE(sendrecv) → 200 OK INVITE(sendrecv)
ISUP/BICC	MGCF	SIP																								
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CPG(retrieve)	→	→ INVITE(recvonly) ← 200 OK INVITE(sendonly)																								
CPG(retrieve)	←	← INVITE(sendrecv) → 200 OK INVITE(sendrecv)																								

TP602006	SIP reference: RFC 3261 [6]	ISUP reference: EN 383 001 [2], annex B.10																								
TSS reference:	ISUP-SIP/SS/HOLD/																									
SIP selection criteria:	Support the temporarily stops sending one or more unicast media streams																									
ISUP selection criteria:	Support the generic notification procedure for HOLD supplementary service																									
Test purpose:	<p>Ensure that a party can put the other party on hold at any time after the call is answered and before call clearing has begun. Ensure that a party can retrieve the call previously put on hold.</p> <ul style="list-style-type: none"> • The calling party should be able to put the other party on hold • The called party should be able to put the other party on hold • The called party should be able to retrieve the other party • The calling party should be able to retrieve the other party 																									
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IAM	→	INVITE																								
ACM	←	180 Ringing																								
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CPG(hold)	←	INVITE(inactive) 200 OK INVITE(inactive)																								
CPG(retrieve)	←	INVITE(recvonly) 200 OK INVITE(sendonly)																								
CPG(retrieve)	→	INVITE(sendrecv) 200 OK INVITE(sendrecv)																								

6.3.2.3 Terminal portability (TP)

TP603001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.13																		
TSS reference:	ISUP-SIP/SS/TP/																			
SIP selection criteria:																				
ISUP selection criteria:	PICS 5/6																			
Test purpose:	<p>Ensure that the SUT stop the temporarily sending one or more unicast media streams if a SUS message (ISDN subscriber initiated) was received.</p> <p>Ensure that the SUT retrieved the media stream if an RES message (ISDN subscriber initiated) was received.</p>																			
SIP Parameter values:	SDP: a=sendonly or a=inactive (suspended) a=sendrecv or a=recvonly or omitted (resumed)																			
ISUP Parameter values:	SUS: Suspend/Resume indicator ISDN subscriber initiated RES: Suspend/Resume indicator ISDN subscriber initiated																			
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">ISUP/BICC</th> <th style="text-align: center; width: 10%;">SUT</th> <th style="text-align: right; width: 60%;">SIP</th> </tr> </thead> <tbody> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">180 Ringing</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">200 OK INVITE</td> </tr> <tr> <td>SUS</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> <tr> <td>RES</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> </tbody> </table>		ISUP/BICC	SUT	SIP	IAM	→	INVITE	ACM	←	180 Ringing	ANM	←	200 OK INVITE	SUS	→	INVITE	RES	→	INVITE
ISUP/BICC	SUT	SIP																		
IAM	→	INVITE																		
ACM	←	180 Ringing																		
ANM	←	200 OK INVITE																		
SUS	→	INVITE																		
RES	→	INVITE																		

TP603002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.13
TSS reference:	ISUP-SIP /SS/TP/	
SIP selection criteria:		
ISUP selection criteria:	PICS 5/6	
Test purpose:	<p>Ensure that the SUT stop the temporarily sending one or more unicast media streams if a SUS message (ISDN subscriber initiated) was received.</p> <p>Ensure that the connection is cleared after T2 was expired in the PSTN.</p>	
SIP Parameter values:	SDP: a=sendonly or a=inactive (suspended)	
ISUP Parameter values:	SUS: Suspend/Resume indicator ISDN subscriber initiated	
Comments:	ISUP/BICC IAM → SUT → SIP INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE SUS → → INVITE T2 expiry REL → → BYE RLC ← ← 200 OK BYE	

6.3.2.4 Conference calling (CONF)

TP604001	SIP reference: RFC 3261 [6]	NGN reference: ES 283 027 [14], clause 7.4.14
TSS reference:	ISUP-SIP/SS/CONF/	
SIP selection criteria:	PICS 8/2	
ISUP selection criteria:	PICS 5/10	
Test purpose:	<p>Ensure that the SUT stop the temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value GEN_NOT_VALUE was received due to the CONF supplementary service.</p> <ul style="list-style-type: none"> If the media stream is either in state "sendonly" or "inactive" then: INVITE with the attribute line a_LINE_VA, or omitted attribute line, else: no mapping. 	
SIP Parameter values:	SDP: a= a_LINE_VA (see table 67) or a line is omitted	
ISUP Parameter values:	CPG: Generic notification = GEN_NOT_VALUE	
Comments:	ISUP/BICC IAM → SUT → SIP INVITE ACM ← ← 180 Ringing ANM ← ← 200 OK INVITE CPG → → INVITE CPG → → INVITE REL → → BYE RLC ← ← 200 OK BYE	

Table 67: Void

TP604003	SIP reference: RFC 3261 [6]	NGN reference: ES 283 027 [14], clause 7.4.14																											
TSS reference:	ISUP-SIP/SS/CONF/																												
SIP selection criteria:	PICS 8/2																												
ISUP selection criteria:	PICS 5/10																												
Test purpose:	<p>Ensure that the SUT stop temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value GEN_NOT_VALUE was received due to the CONF supplementary service.</p> <ul style="list-style-type: none"> If the media stream is either in state "sendonly" or "inactive" then: INVITE with the attribute line a_LINE_VA, or omitted attribute line, else: no mapping. 																												
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ISUP/BICC	SUT	SIP																											
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CPG	→	INVITE																											
CPG	→	INVITE																											
REL	→	BYE																											
RLC	←	200 OK BYE																											

Table 68: Void

TP604005	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.14 1.7/Q.734																														
TSS reference:	ISUP-SIP/SS/CONF/																															
SIP selection criteria:																																
ISUP selection criteria:	NOT PICS 5/10																															
Test purpose:	<p>Ensure that the SUT on receipt of a CPG message due to the CONF supplementary service, the Generic notification indicator with the value.</p> <p>No mapping, no disrupting the SIP procedure.</p>																															
SIP Parameter values:	No mapping																															
ISUP Parameter values:	CPG: Generic notification = Conference established CPG: Generic notification = isolated CPG: Generic notification = reattached CPG: Generic notification = Conference disconnected																															
Comments:	<table style="width: 100%; border-collapse: collapse;"> <thead> <tr> <th style="text-align: left; width: 30%;">ISUP/BICC</th> <th style="text-align: center; width: 10%;">SUT</th> <th style="text-align: right; width: 60%;">SIP</th> </tr> </thead> <tbody> <tr> <td>IAM</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> <tr> <td>ACM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">180 Ringing</td> </tr> <tr> <td>ANM</td> <td style="text-align: center;">←</td> <td style="text-align: right;">200 OK INVITE</td> </tr> <tr> <td>CPG</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> <tr> <td>CPG</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> <tr> <td>CPG</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> <tr> <td>CPG</td> <td style="text-align: center;">→</td> <td style="text-align: right;">INVITE</td> </tr> <tr> <td>REL</td> <td style="text-align: center;">→</td> <td style="text-align: right;">BYE</td> </tr> <tr> <td>RLC</td> <td style="text-align: center;">←</td> <td style="text-align: right;">200 OK BYE</td> </tr> </tbody> </table>	ISUP/BICC	SUT	SIP	IAM	→	INVITE	ACM	←	180 Ringing	ANM	←	200 OK INVITE	CPG	→	INVITE	REL	→	BYE	RLC	←	200 OK BYE										
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IAM	→	INVITE																														
ACM	←	180 Ringing																														
ANM	←	200 OK INVITE																														
CPG	→	INVITE																														
CPG	→	INVITE																														
CPG	→	INVITE																														
CPG	→	INVITE																														
REL	→	BYE																														
RLC	←	200 OK BYE																														

6.3.2.5 Three Party service (3PTY)

TP605001	SIP reference: RFC 3261 [6]	NGN reference: ES 283 027 [14], clause 7.4.15																																				
TSS reference:	ISUP-SIP/SS/3PTY/																																					
SIP selection criteria:	PICS 8/2																																					
ISUP selection criteria:	PICS 5/5 AND PICS 5/18																																					
Test purpose:	<p>Ensure that the SUT stop the temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value GEN_NOT_VALUE was received due to the CONF supplementary service.</p> <ul style="list-style-type: none"> If the media stream is either in state "sendonly" or "inactive" then: INVITE with the attribute line a_LINE_VA, or omitted attribute line, else: no mapping. 																																					
SIP Parameter values:	SDP: a= a_LINE_VA (see table 69) or a line is omitted																																					
ISUP Parameter values:	CPG: Generic notification = remote hold CPG: Generic notification = GEN_NOT_VALUE																																					
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP/BICC</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">SIP</td> <td style="width: 25%;"></td> </tr> <tr> <td>IAM</td> <td>→</td> <td>→</td> <td>INVITE</td> </tr> <tr> <td>ACM</td> <td>←</td> <td>←</td> <td>180 Ringing</td> </tr> <tr> <td>ANM</td> <td>←</td> <td>←</td> <td>200 OK INVITE</td> </tr> <tr> <td>CPG</td> <td>→</td> <td>→</td> <td>INVITE</td> </tr> <tr> <td>CPG</td> <td>→</td> <td>→</td> <td>INVITE</td> </tr> <tr> <td>CPG</td> <td>→</td> <td>→</td> <td>INVITE</td> </tr> <tr> <td>REL</td> <td>→</td> <td>→</td> <td>BYE</td> </tr> <tr> <td>RLC</td> <td>←</td> <td>←</td> <td>200 OK BYE</td> </tr> </table>	ISUP/BICC	SUT	SIP		IAM	→	→	INVITE	ACM	←	←	180 Ringing	ANM	←	←	200 OK INVITE	CPG	→	→	INVITE	CPG	→	→	INVITE	CPG	→	→	INVITE	REL	→	→	BYE	RLC	←	←	200 OK BYE	
ISUP/BICC	SUT	SIP																																				
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CPG	→	→	INVITE																																			
REL	→	→	BYE																																			
RLC	←	←	200 OK BYE																																			

TP605002	SIP reference: RFC 3261 [6]	NGN reference: ES 283 027 [14], clause 7.4.15																																
TSS reference:	ISUP-SIP/SS/3PTY /																																	
SIP selection criteria:	PICS 8/1																																	
ISUP selection criteria:	PICS 5/5 AND PICS 5/18																																	
Test purpose:	<p>Ensure that the SUT stop the temporarily sending one or more unicast media streams if a CPG message Generic notification indicator with the value GEN_NOT_VALUE was received due to the CONF supplementary service in the ALERTING state.</p> <ul style="list-style-type: none"> If the media stream is either in state "sendonly" or "inactive" then: INVITE with the attribute line a_LINE_VA, or omitted attribute line, else: no mapping. 																																	
SIP Parameter values:	SDP: a= a_LINE_VA (see table 69) or a line is omitted																																	
ISUP Parameter values:	CPG: Generic notification = remote hold CPG: Generic notification = GEN_NOT_VALUE																																	
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP/BICC</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">SIP</td> <td style="width: 25%;"></td> </tr> <tr> <td>IAM</td> <td>→</td> <td>→</td> <td>INVITE</td> </tr> <tr> <td>ACM</td> <td>←</td> <td>←</td> <td>180 Ringing</td> </tr> <tr> <td>CPG</td> <td>→</td> <td>→</td> <td>UPDATE</td> </tr> <tr> <td>CPG</td> <td>→</td> <td>→</td> <td>UPDATE</td> </tr> <tr> <td>CPG</td> <td>→</td> <td>→</td> <td>UPDATE</td> </tr> <tr> <td>REL</td> <td>→</td> <td>→</td> <td>BYE</td> </tr> <tr> <td>RLC</td> <td>←</td> <td>←</td> <td>200 OK BYE</td> </tr> </table>	ISUP/BICC	SUT	SIP		IAM	→	→	INVITE	ACM	←	←	180 Ringing	CPG	→	→	UPDATE	CPG	→	→	UPDATE	CPG	→	→	UPDATE	REL	→	→	BYE	RLC	←	←	200 OK BYE	
ISUP/BICC	SUT	SIP																																
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CPG	→	→	UPDATE																															
REL	→	→	BYE																															
RLC	←	←	200 OK BYE																															

Table 69

Values for test purpose TP605001, TP605002		
CPG→ Generic notification GEN_NOT_VALUE		INVITE/UPDATE→ SDP attribute line a_LINE_VA
VA_01	Conference established	sendrecv, or recvonly
VA_02	Conference disconnected	sendrecv or recvonly

TP605003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.15 2.7/Q.734
TSS reference:	ISUP-SIP/SS/3PTY/	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 5/18	
Test purpose:	Ensure that the SUT on receipt of a CPG message due to the 3PTY supplementary service, the Generic notification indicator with the value. No mapping, no disrupting the SIP procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	CPG: Generic notification = remote hold CPG: Generic notification = Conference established CPG: Generic notification = Conference disconnected	
Comments:	ISUP/BICC IAM → ACM ← ANM ← CPG → CPG → CPG → REL → RLC ←	SUT → INVITE 180 Ringing 200 OK INVITE BYE 200 OK BYE

6.3.2.6 Connected line identification (COL)

TP606001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.2
TSS reference:	ISUP-SIP/SS//COL /	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if the IAM is received with an optional forward call indicator, connected line requested, continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:		
	ISUP IAM → ACM ← ANM ← REL → RLC ←	SUT Conversation → INVITE ← 180 Ringing ← 200 OK INVITE → ACK → BYE ← 200 OK BYE

6.3.2.7 Sub-addressing (SUB)

TP607001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.5
TSS reference:	ISUP-SIP/SS/ SUB /	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if the IAM is received with an ATP containing a calling sub-address, continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM → ACM ← ANM ← REL → RLC ←	SUT Conversation → INVITE ← 180 Ringing ← 200 OK INVITE → BYE ← 200 OK BYE

6.3.2.8 Closed user group (CUG)

TP608001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.16
TSS reference:	ISUP-SIP/SS/CUG/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if an IAM is received with Optional forward call indicator, CUG call indicator coded as " CUG call with outgoing access " and CUG interlock code or CUG call indicator coded as "Non CUG call" or Optional forward call indicator is absent, the SIP signalling procedure is not disrupted.	
SIP Parameter values:	No mapping	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM → ACM ← ANM ← REL → RLC ←	SUT → INVITE ← 180 Ringing ← 200 OK INVITE Conversation → Conversation ← BYE → 200 OK BYE

TP608002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.16
TSS reference:	ISUP-SIP/SS/CUG/	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if an IAM is received with Optional forward call indicator, CUG call indicator coded as " CUG call without outgoing access " and CUG interlock code, a REL is sent. No INVITE is sent into the SIP network.	
SIP Parameter values:	No action	
ISUP Parameter values:	REL: Cause #29	
Comments:	ISUP/BICC IAM → REL ← RLC →	SUT → SIP

6.3.2.9 Call diversion (CDIV)

TP609001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annexes B.6 and B.7
TSS reference:	ISUP-SIP/SS/ CDIV /	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if the IAM is received with Redirecting number, original called number and redirection information , continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	IAM: Redirecting number, Original called number, Redirection information	
Comments:	ISUP/BICC IAM → SUT → SIP ACM ← ← INVITE ANM ← ← 180 Ringing REL → Conversation → 200 OK INVITE RLC ← 200 OK BYE	200 OK INVITE BYE 200 OK BYE

6.3.2.10 User to user signalling (UUS)

TP610001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.1.7/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:		
Test purpose:	Ensure that the SUT if the IAM is received with User-to-user information as an implicit service 1 request returns a User-to-user indicator in the ACM "UUI discarded by the network" and continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: User-to-indicator "UUI discarded by the network", Service 1 response "No indication", BCI: "Interworking encountered".	
Comments:	ISUP/BICC IAM → SUT → SIP ACM ← ← INVITE ANM ← ← 180 Ringing REL → Conversation → 200 OK INVITE RLC ← 200 OK BYE	200 OK INVITE BYE 200 OK BYE

TP610002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.1.7/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	Ensure that the SUT if the IAM is received with an explicit service 1 request "Not essential" returns a User-to-user indicator in the ACM "Service 1 not provided" and continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: User-to-indicator "UUI discarded by the network", Service 1 response "Nt provided"	
Comments:	ISUP/BICC IAM ACM ANM REL RLC	SUT → ← ← Conversation → ← SIP INVITE 180 Ringing 200 OK INVITE Conversation → BYE 200 OK BYE

TP610003	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.1.7/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	Ensure that the SUT if the IAM is received with an explicit service 1 request "essential" returns a REL with cause #29 and an diagnostics containing the user-to-user indicator parameter name.	
SIP Parameter values:	No action	
ISUP Parameter values:	REL: cause #29, diagnostics value 0x2a	
Comments:	ISUP/BICC IAM REL RLC	SUT → ← SIP

TP610004	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.2.7/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	Ensure that the SUT if the IAM is received with an explicit service 2 request "Not essential" returns a User-to-user indicator in the ACM "Service 2 not provided" and continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: User-to-indicator Service 2 response "Not provided"	
Comments:	ISUP/BICC IAM ACM ANM REL RLC	SUT → ← ← Conversation → ← SIP INVITE 180 Ringing 200 OK INVITE Conversation → BYE 200 OK BYE

TP610005	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.2.7/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	Ensure that the SUT if the IAM is received with an explicit service 2 request "essential" returns a REL with cause #29 and an diagnostics containing the user-to-user indicator parameter name.	
SIP Parameter values:	No action	
ISUP Parameter values:	REL: cause #29, diagnostics value 0x2a	
Comments:	ISUP/BICC IAM REL RLC	SUT → ← SIP

TP610006	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.3.7.1/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	Ensure that the SUT if the IAM is received with an explicit service 3 request "Not essential" returns a User-to-user indicator in the ACM "Service 1 not provided" and continue without disrupting the SIP or ISUP signalling procedure.	
SIP Parameter values:	No mapping	
ISUP Parameter values:	ACM: User-to-indicator, Service 3 response "Not provided"	
Comments:	ISUP/BICC IAM ACM ANM REL RLC	SUT → ← ← Conversation → ← SIP INVITE 180 Ringing 200 OK INVITE Conversation → BYE 200 OK BYE

TP610007	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.3.7.1/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	Ensure that the SUT if the IAM is received with an explicit service 3 request "essential" returns a REL with cause #29 and an diagnostics containing the user-to-user indicator parameter name.	
SIP Parameter values:	No action	
ISUP Parameter values:	REL: cause #29, diagnostics value 0x2a	
Comments:	ISUP/BICC IAM REL RLC	SUT → ← SIP

TP610008	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.3.7.2/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	PICS 11/1 AND PICS 11/2	
Test purpose:	Ensure that the SUT if the FAR is received with an explicit service 3 request "Not essential" returns a FRJ with cause #29.	
SIP Parameter values:	No action	
ISUP Parameter values:	FRJ: User-to-user indicator = "Service 3 not provided"	
Comments:	ISUP/BICC IAM → SUT → SIP ACM ← → INVITE ANM ← → 180 Ringing Conversation ← → 200 OK INVITE FAR → Conversation FRJ ← Conversation REL → Conversation → BYE RLC ← → 200 OK BYE	

TP610009	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.1.5.2.5.2.2/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	Ensure that the SUT if the IAM is received with an explicit service 1 request "Not essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No mapping	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM → SUT → SIP ACM ← → INVITE ANM ← → 180 Ringing Conversation ← → 200 OK INVITE REL → Conversation → BYE RLC ← → 200 OK BYE	

TP610010	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.1.5.2.5.2.2/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	Ensure that the SUT if the IAM is received with an explicit service 1 request "essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No action	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM ACM ANM REL RLC	SUT → ← ← Conversation → ← → SIP INVITE 180 Ringing 200 OK INVITE BYE 200 OK BYE

TP610011	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.2.5.2.5.2.1/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	Ensure that the SUT if the IAM is received with an explicit service 2 request "Not essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No mapping	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM ACM ANM REL RLC	SUT → ← ← Conversation → ← → SIP INVITE 180 Ringing 200 OK INVITE BYE 200 OK BYE

TP610012	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.2.5.2.5.2.1/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	Ensure that the SUT if the IAM is received with an explicit service 2 request "essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No action	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM ACM ANM REL RLC	SUT → ← ← Conversation → ← SIP INVITE 180 Ringing 200 OK INVITE Conversation → BYE 200 OK BYE

TP610013	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.3.5.2.5.2.1/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	Ensure that the SUT if the IAM is received with an explicit service 3 request "Not essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No mapping	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM ACM ANM REL RLC	SUT → ← ← Conversation → ← SIP INVITE 180 Ringing 200 OK INVITE Conversation → BYE 200 OK BYE

TP610014	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.3.5.2.5.2.1/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/2	
Test purpose:	Ensure that the SUT if the IAM is received with an explicit service 3 request "essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No action	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM ACM ANM REL RLC	SUT → ← ← Conversation → ← → SIP INVITE 180 Ringing 200 OK INVITE BYE 200 OK BYE

TP610015	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.21 1.3.5.2.5.2.1/Q.737
TSS reference:	ISUP-SIP/SS/ UUS /	
SIP selection criteria:		
ISUP selection criteria:	NOT PICS 11/1 OR NOT PICS 11/3	
Test purpose:	Ensure that the SUT if the FAR is received with an explicit service 3 request "Not essential" continue without disrupting the SIP or ISUP signalling procedure. No response to this request.	
SIP Parameter values:	No action	
ISUP Parameter values:		
Comments:	ISUP/BICC IAM ACM ANM FAR REL RLC	SUT → ← ← Conversation → Conversation → → → SIP INVITE 180 Ringing 200 OK INVITE Conversation Conversation → BYE 200 OK BYE

6.3.2.11 Explicit call transfer (ECT)

TP611001	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.8																																	
TSS reference:	ISUP-SIP/SS/ECT/																																		
SIP selection criteria:																																			
ISUP selection criteria:	PICS 12/1																																		
Test purpose:	<p>Ensure that the SUT if a LOP(request) is received returns a LOP (response) with the indication "insufficient information" continue without disrupting the SIP signalling procedure.</p> <p>Ensure that the SUT if a FAC is received continue without disrupting the SIP signalling procedure.</p>																																		
SIP Parameter values:	No mapping																																		
ISUP Parameter values:	LOP: Response "insufficient information"																																		
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP/BICC</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">SIP</td> </tr> <tr> <td>IAM</td> <td>→</td> <td>INVITE</td> </tr> <tr> <td>ACM</td> <td>←</td> <td>180 Ringing</td> </tr> <tr> <td>ANM</td> <td>←</td> <td>200 OK INVITE</td> </tr> <tr> <td></td> <td>Conversation</td> <td>Conversation</td> </tr> <tr> <td>LOP</td> <td>→</td> <td></td> </tr> <tr> <td>LOP</td> <td>←</td> <td></td> </tr> <tr> <td>FAC</td> <td>→</td> <td></td> </tr> <tr> <td></td> <td>Conversation</td> <td>Conversation</td> </tr> <tr> <td>REL</td> <td>→</td> <td>BYE</td> </tr> <tr> <td>RLC</td> <td>←</td> <td>200 OK BYE</td> </tr> </table>	ISUP/BICC	SUT	SIP	IAM	→	INVITE	ACM	←	180 Ringing	ANM	←	200 OK INVITE		Conversation	Conversation	LOP	→		LOP	←		FAC	→			Conversation	Conversation	REL	→	BYE	RLC	←	200 OK BYE	
ISUP/BICC	SUT	SIP																																	
IAM	→	INVITE																																	
ACM	←	180 Ringing																																	
ANM	←	200 OK INVITE																																	
	Conversation	Conversation																																	
LOP	→																																		
LOP	←																																		
FAC	→																																		
	Conversation	Conversation																																	
REL	→	BYE																																	
RLC	←	200 OK BYE																																	

TP611002	SIP reference: RFC 3261 [6]	ISUP reference: ITU-T Rec Q.1912.5 [1], annex B.8																														
TSS reference:	ISUP-SIP/SS/ECT/																															
SIP selection criteria:																																
ISUP selection criteria:	NO PICS 12/1																															
Test purpose:	<p>Ensure that the SUT if a LOP(request) is received continue without disrupting the SIP signalling procedure.</p> <p>Ensure that the SUT if a FAC is received continue without disrupting the SIP signalling procedure.</p>																															
SIP Parameter values:	No mapping																															
ISUP Parameter values:																																
Comments:	<table style="width: 100%; border-collapse: collapse;"> <tr> <td style="width: 25%;">ISUP/BICC</td> <td style="width: 25%; text-align: center;">SUT</td> <td style="width: 25%; text-align: center;">SIP</td> </tr> <tr> <td>IAM</td> <td>→</td> <td>INVITE</td> </tr> <tr> <td>ACM</td> <td>←</td> <td>180 Ringing</td> </tr> <tr> <td>ANM</td> <td>←</td> <td>200 OK INVITE</td> </tr> <tr> <td></td> <td>Conversation</td> <td>Conversation</td> </tr> <tr> <td>LOP</td> <td>→</td> <td></td> </tr> <tr> <td>FAC</td> <td>→</td> <td></td> </tr> <tr> <td></td> <td>Conversation</td> <td>Conversation</td> </tr> <tr> <td>REL</td> <td>→</td> <td>BYE</td> </tr> <tr> <td>RLC</td> <td>←</td> <td>200 OK BYE</td> </tr> </table>	ISUP/BICC	SUT	SIP	IAM	→	INVITE	ACM	←	180 Ringing	ANM	←	200 OK INVITE		Conversation	Conversation	LOP	→		FAC	→			Conversation	Conversation	REL	→	BYE	RLC	←	200 OK BYE	
ISUP/BICC	SUT	SIP																														
IAM	→	INVITE																														
ACM	←	180 Ringing																														
ANM	←	200 OK INVITE																														
	Conversation	Conversation																														
LOP	→																															
FAC	→																															
	Conversation	Conversation																														
REL	→	BYE																														
RLC	←	200 OK BYE																														

Annex A (informative): Bibliography

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History

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